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THE INTELLIGIBILITY OF WORDS, SENTENCES, AND CONTINUOUS DISCOURSE, USING THE ARTICULATION INDEX

by

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The purpose of this research was to investigate the effect of message redundancy upon intelligibility. The original methodology for the Articulation Index (AI) [French and Steinberg, J. Acoust. Soc. Am., 19, 90-119, 1947] was used to examine the relation between words, meaningful sentences and continuous discourse (CD). One primary consideration was to derive the relations between the three speech types with tightly controlled, highly repeatable experimental conditions such that any difference between them could be attributed solely to inherent contextual differences.

One male speaker recorded 616 monosyllabic words, 176 meaningful speech perception in noise (SPIN) sentences and 44 seventh-grade reading level CD passages. Twenty-four normal hearing subjects made intelligibility estimates of the CD and sentences and identified words at each of 44 conditions of filtering and signal-to-noise ratio. The sentence intelligibility scores and continuous discourse intelligibility scores plotted versus the AI (transfer function) were within 0.05 AI of each other. The word recognition scores were considerably lower for equivalent AI values of both sentences and CD.

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ABSTRACT

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One male speaker recorded 616 monosyllabic words, 176 meaningful speech perception in noise (SPIN) sentences and 44 seventh-grade reading level CD passages. Twenty-four normal hearing subjects made intelligibility estimates of the CD and sentences and identified words at each of 44 conditions of filtering and signal-to-noise ratio. The sentence intelligibility scores and continuous discourse intelligibility scores plotted versus the AI (transfer function) were within 0.05 AI of each other. The word recognition scores were considerably lower for equivalent AI values of both sentences and CD.

A plot of the importance of the frequency bands used in this study toward understanding speech revealed that the area of most importance was centered around 2000 Hz for all three types of speech. As message redundancy increased (words to sentences to CD) the shape of this area spread progressively to include lower and higher frequencies. A recalculation of the frequency importance function into bands comparable to octave bands revealed that the sentence and CD functions were nearly identical. This fact coupled with the similarity of the sentence and CD transfer functions implies that the two speech types can be used interchangeably when computing the octave band AI. However, the differences between the frequency importance functions in the smaller bands used in

this study demonstrated that the assumption that one frequency importance function can be used to compute the AI for all different types of speech is not valid.

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Chapter 1

INTRODUCTION

A considerable amount of research has been published on different aspects of message redundancy and their relations to speech intelligibility. Reported studies have been concerned with speech at various levels of its complexity; words, sentences and continuous discourse (CD). Some researchers have concentrated solely upon techniques to measure aspects of intelligibility for these three types of speech. The result of their efforts have been well established methods for assessing the intelligibility (or some aspect of intelligibility) for words (ANSI S3.2, 1991, Tillman, Carhart and Wilbur, 1963 and Tillman and Carhart, 1966), sentences (Kalikow, 1977, Duffy and Giolas, 1974 and Giolas, Cooker and Duffy, 1970) and CD (Speaks, Parker and Kuhl, 1972 and Giolas, 1966).

Other researchers have used the developed methods to try to understand how speech with varying message redundancy is perceived. For example, some researchers have tried to isolate syntactic and semantic cues in sentences and CD (Lea, 1973 and Miller, 1962). In other studies attempts have been made to isolate some aspect of word intelligibility such as the effects of word frequency upon word recognition (Pollack, Rubenstein and Decker, 1959 and Pisoni, Nusbaum, Luce and Slowiaczek, 1985).

A very important question in addressing message redundancy is how these three basic types of speech differ from one another in the speech perception process. One important way to observe these differences is to analyze speech perception in a background of noise. Words are not understood as well as sentences or CD when presented in an identical background of noise (Miller, Heise and Lichten, 1951). The amount of information contained within each letter comprising a word also changes dramatically as the written

sample of speech spans more than one word (Shannon, 1951 and Burton and Licklidder, 1955). Further, the identification of missing words within a sentence is highly dependent upon the semantic cues within the sentence (Giolas et al., 1970). In addition, it seems that listeners use different stategies to perceive speech depending on their motivation, expectations, ability to concentrate, etc.

One measurable aspect of speech perception is the way that listeners use available acoustic information to understand speech. Although the use of acoustic information by listeners has been studied extensively for single types of speech (Miller and Nicely, 1951; Black, 1959; Duggirala, Studebaker, Pavlovic and Sherbecoe, 1988; French and Steinberg, 1947 and Studebaker, Pavlovic and Sherbecoe, 1987), very little research has concentrated upon this relation for words, sentences and CD within a single study. The comparison between studies invariably leads to problems, since different stimuli, methodology and subjects are used in different studies.

The Articulation Index (AI) is an excellent tool for examining parameters of speech intelligibility. The AI is a measure between zero and one, of the amount of acoustic information that is available at the listener's ear(s). The underlying concept of the AI is that speech intelligibility is proportional to the average difference in dB between the masking level of the noise and the long term root mean square (rms) of the speech material plus the level of the speech peaks above the long term rms. The AI also can be used to derive a speech frequency importance function, which defines how important different portions of the speech spectrum are for speech understanding. The research reported here investigates the intelligibility of three types of speech (words, sentences and CD) using the AI as a criterion. A unique aspect of this research is its use of one talker and one set of listeners to examine all three types of speech. Therefore, the final conclusions may be drawn with no intra study bias.

Chapter 2

BACKGROUND

Intelligibility and Context

Limited Stimulus and Repetitive Messages-Words

Limiting the contextual cues in a speech stimulus clearly influences recognition of the speech stimulus. For example, Miller et al. (1951) found that at the same signal to noise ratio (S/N ratio), the listeners had articulation scores of 100, 80 and 40% for digits, sentences and nonsense syllables respectively. In part, this occurred because digits are constrained to only nine possible choices and distinguished simply by recognizing the vowel (except five and nine). Seven also has two syllables, while all others have only one syllable. At the other extreme, the nonsense syllables presented the listeners with a large number of choices, no choice more probable than any other. Yet, the sole reason for increased recognition of restricted sets of speech material is not merely the reduction of the actual set of words to be identified. That is, the number of distinct speech sounds which must be discriminated is also a controlling factor. The nonsense syllable scores were considerably lower than sentence scores because they have no contextual or linguistic restrictions upon speech sounds that follow one another; whereas a meaningful sentence places syntactic and semantic structure upon the words within it. Thus, nonsense syllables place no limit on the probable pool of sounds following one another, while digits place severe limits upon the pool of speech sounds.

Pollack et al. (1959) studied speech intelligibility in known and unknown message sets. A known message set contained 8 words while the unknown message set contained 144 words. The words were from eight different classes of word frequency. For known

message sets, the primary factor controlling the intelligibility scores was the confusedness of words within a set. For unknown message sets it was the frequency of occurrence of the word, relative to the frequency of occurrence of words with which a stimulus word might be confused, that mattered. Unlike the study by Miller et al. (1951) that used digits, Pollack et al. (1959) introduced frequency of occurrence as another variable. Thus, the number of distinct speech sounds to be discriminated, as well as the potential for confusing each word with other frequently occurring words, must be accounted for in studies of speech intelligibility. Further, using known and unknown message sets brings up an important question. Can an unknown message set of nonsense syllables be used as a test of speech intelligibility? Speech intelligibility infers understanding, which in turn assumes that the stimulus has meaning. Nonsense syllables have no meaning, so their identification could be classified as a speech recognition task. In meaningful speech, the characterization of phonemes is not solely by their phonetic properties; they are also characterized by their position within an utterance and the distribution of that phoneme within the sequence of spoken language (Lehiste and Peterson, 1959).

Shannon (1951) developed a method to investigate the dependence of linguistic restrictions on the structure of words. He presented subjects with text material. The subject's task was to guess a letter based upon the preceding letters. Shannon found that the bits of information ($\log_2$/probability of the letter occurring) contained in each letter within a word depended upon its position in that word. For example, in a typical five letter word, the first letter contained 4.8 bits versus the last letter, which contained 2.3 bits of information. As such there was a definite relation between letters. Since nonsense syllables do not exploit this relation, they clearly do not utilize contextual information.

The effect of limiting stimuli on speech intelligibility is highly dependent upon the number of words that the listener could potentially confuse with the stimulus word. The

confusedness of certain types of words is also varied. For example, as the syllable length increases the intelligibility scores for each word also increase (Egan, 1948). One reason is that as syllable length increases there is a decrease in the pool of items that the word could be confused with. Two words, cat and symphony, illustrate this point. While there is an enormous pool of words similar to cat, symphony is not easily confused with many words. It seems that the brain, unlike a computer, is very good at making slow decisions about a large amount of information. Reducing the quantity of this information by increasing syllable length, increases the chances of correctly perceiving a word.

The repetition of test items also seems one way to improve intelligibility scores. Yet, Miller et al. (1951) found just the opposite. Repeating monosyllabic words three times only produced a small increase (5-10%) in scores from the first to the second presentation. There was no increase in scores from the second to the third presentation. Apparently, since subsequent presentations of the same material supply no new information, the listener cannot narrow the range of possible words. Thwing (1956) confirmed this finding. On the other hand, Clark et al. (1985) presented some conflicting data. In contrast, they observed a significant increase of 18.6% in speech intelligibility between the first and second presentations of consonant-vowel (CV) nonsense syllables. The repetition of nonsense syllables can conceivably add some information since nonsense syllables have minimum phonological structure.

Context and Semantic Cues--Sentences

Semantic cues for speech intelligibility are supplied by both the structure and the meaning of the language. However, it is difficult to separate meaning from structure, syntax and semantics. For example, the sentence, "Harry sleeps in a ______________." cannot be completed with a verb. The answer must be a noun that can describe what was slept in. Here the syntax is as important as the semantics.

Miller et al. (1951) presented subjects with sentences containing five key words. The same key words were also presented randomly in isolation to other subjects. The key words in sentences produced articulation scores that were considerably higher than the word in isolation scores. At an SNR of +6 dB the articulation score for the key words in sentences was 90% compared with 60% for the same key words in isolation. The effect of adding semantic cues upon intelligibility is similar to increasing the syllable length of a word. The possible choices are narrowed in both cases. Unfortunately, semantic cues are hard to quantify since variables such as intelligence and memory enter into the task (Giolas, 1966).

Burton and Licklidder (1955) used the same guessing technique as Shannon (1951) to find the point at which the preceding information in text does not add any new information. They determined this point to be at 32 letters or about five or six words. In other words, the introduction of more than five or six words does not facilitate the guessing of the next letter. Though Burton and Licklidder (1955) used guessing of written text, the results offer some insights into spoken communication. Guessing succeeding letters in text involves the use of semantic and syntactic factors, possibly the same processing that is used to understand meaningful speech. Whether or not this is true, since a short sentence is about five or six words, this study does imply that the structure of a sentence is an efficient method of communicating an idea.

The structure of the sentence is another potential variable for speech intelligibility. For example, if a key word in a sentence is missing, but the key word can still be determined, the word predictability is high for that key word. Giolas et al. (1970) studied word predictability for different types of sentence constructions. They used a procedure of randomly eliminating words in sets of sentences. The total number of words eliminated in any set of sentences was controlled. The results varied widely according to the contextual

clues present in the sentences. Synthetic sentences (Jerger et al., 1968), Central Institute for the Deaf (CID) sentence lists B and D, and CID revised sentence list C (Jerger et al., 1968) were used as test stimuli. The synthetic sentences contained very few contextual clues and the CID list B sentences contained many contextual clues. The identification of words in the synthetic sentences did not vary as a function of the number of words removed. This was expected since the synthetic sentences present a task similar to guessing a word with only syntax as a clue. However, the CID list B sentences, which are replete with contextual clues, had intelligibility ranging from 70% with 10 words missing to 30% with 25 words missing. This study demonstrated the danger of generalizing stimulus specific research to other studies. Even if an intelligibility study uses sentences as the main stimulus, the structure of these sentences might render the study incomparable to research using different sentences.

Duffy and Giolas (1974) studied the effect of word predictability upon sentence intelligibility. They used the CID lists from the previous study to represent a wide range of word predictability. To degrade the stimulus enough to compare errors, low pass filters of 420 and 360 Hz were used. The subjects wrote the sentence down after hearing it. The results demonstrated the relationship between high predictability words and high sentence articulation scores. In other words, the structure of a sentence and its context can have a large influence upon intelligibility.

The problem of developing sets of sentences with controlled word predictability and of equal difficulty was addressed by Kalikow et al. (1977). The major objective in this research was "to produce a measure that would assess the utilization of linguistic-situational information in comparison with utilization of acoustic phonetic information" (Kalikow et al., 1977, p. 1339). They did this by developing 10 forms of 50 sentences where half the forms contained meaningful and the other half contained non-meaningful

sentences. The test was called the Speech Perception In Noise (SPIN) test. The sentences were designed to elicit a one word (key word) response, that being the last word in the sentence. The sentences were all five to eight words in length with a key word that was a monosyllabic noun with a frequency count of 5 to 150 per million words. The predictability of the key words in the contextual sentences was determined by presenting those sentences to subjects without the key word. The subject was then asked to write down the word most likely to occur. Once developed, equivalence and phonetic balance of the forms was tested.

One of the original purposes of the SPIN test was to determine the subject's use of contextual information. This was achieved by dividing the 50 sentence forms into two parts. One part contained 25 meaningful sentences and the other contained 25 non-meaningful sentences. The difference between these two parts was called the difference score and should yield information about how a subject uses context. Owen (1981) studied the relations of the difference score to syntactic skills, semantic skiils, intelligence quotient, hearing loss, and signal to noise (S/N) ratio. He found the difference score related mostly to the subject's hearing and the S/N ratio and concluded that the difference score was not an effective measure of a subject's use of contextual information. There was also some question of the equivalence of the sentence forms. Equivalence is necessary if the forms are to be used interchangeably (one forms score considered equivalent to another forms score in the same condition of noise or distortion). Morgan et al. (1981) found that three of the ten lists were suspect. Bilger (1985) refined the test into eight equivalent forms that he renamed the revised SPIN test.

Context and Semantic Cues--Continuous Discourse

The ideal method for assessing a person's ability to understand speech would be to use speech material that represents everyday communication. The most natural choice of

speech stimuli would be continuous discourse (CD). However, reliable, quick and repeatable methods for using CD to measure aspects of speech intelligibility are very difficult to achieve. There is also a problem with the content of a CD passage. A passage that challenges a subject intellectually will not effectively assess the subject's perception of the speech, nor will it do an effective job of assessing a communication system (Giolas, 1966).

Attempts have been made to use CD as a test of speech intelligibility. Giolas (1966) compared the intelligibility of CID isolated sentences to a fifteen minute CD passage under seven filter conditions. Except at low pass frequencies below 1260 Hz, there was almost no difference between the intelligibility of CD and CID sentences. That CD can be understood when only low frequencies are present could be significant in some environments.

The decrease in the importance of high frequencies to intelligibility as context increases has been demonstrated in other studies (Pollack and Pickett, 1964; Studebaker et al., 1987). That is, as the redundancy of the stimulus increases there is a trend that lower frequencies become more important. One explanation is that as the contextual clues increase, less overall acoustic information is needed to process speech. Instead, semantic and syntactical cues are used more for perception. In particular, the need to understand the high frequency cues of the consonants becomes less important.

In order to understand the effects of contextual cues upon CD, it is necessary to have some reliable method for quantifying the intelligibility of CD passages. The approach used by Giolas (1966) has the disadvantage of measuring the subject's memory and intelligence and provides at best, an indirect measurement of the amount of the message understood. Another method is to have the subject repeat back what they hear, or to shadow the passage. The major drawback of this method is that it involves two separate processes, a perceptual process and the motor skills involved in speaking.

Speaks et al. (1972) developed an estimation technique. Performance intensity functions were developed for subjects using a constant percentage criteria and for subjects estimating the percentage of material they understood. The results show that if the purpose is to measure how well a CD passage is understood and not to measure the types of errors, the estimation of the percentage understood by the subject is a viable method. This conclusion was also reached by Cox et al. (1991). They compared a sentence repetition task to a sentence estimation task for conversational speech and found no significant difference for young normal hearing listeners.

Summary

The discussion above, of the various factors that affect the intelligibility of speech, shows that a study involving speech must have very extensive and not always easy to grasp controls. Two experiments with identical methodology and different speech stimuli are not identical. Even if both experiments use sentences, the structure and the meaning contained within the sentences also must be identical to afford any comparison.

Further, difficulty in controlling intelligibility factors increases as the message redundancy increases. A word based intelligibility test, at the very least, is biased by the syllable length of the word, the frequency of occurrence of the word, and the subject's knowledge of the stimulus. In a sentence intelligibility task, these factors as well as semantic factors are involved. Also, care must be taken not to confound results beause of subject variables such as intelligence and memory. A well designed speech intelligibility experiment should contain well tested stimuli with as much control upon syntactic and semantic cues as is currently possible.

The Articulation Index

There are many different methods of controlling contextual clues for words and sentences. Still, beyond the control of contextual cues, some type of paradigm is needed to measure the effect of context upon speech intelligibility. There are two procedures for predicting the intelligibility of speech in a particular environment. These are the Speech Transmission Index (STI) and the Articulation Index (AI). Both have proven effective for predicting speech recognition scores of listeners in noisy situations (Kryter, 1962b, Steeneken and Houtgast, 1980). The STI uses a modulated signal to predict the smearing of the speech signal and thus might have an advantage over the AI in environments with reverberation. For this reason, the STI has proven very effective in room and hall acoustics (Houtgast and Steeneken, 1985). The method for deriving the AI (French and Steinberg, 1947) using high and low pass filters, makes it ideal for investigating the frequency dependence of different speech material.

Historical Aspects of AI

The study of speech intelligibility has led to many theories predicting the effectiveness of communication systems. From a practical viewpoint, it would be very cost efficient to predict the success of a communication system before implementing it. The AI was developed at Bell Laboratoriess for assessing telephony (French and Steinberg, 1947; Fletcher and Galt, 1950). Spurred on by research that was performed before and during World War II, the AI was based on four studies examining the relations between articulation scores, different SNRs and low and high pass filtering. The goal was to identify the most important parameters that effect speech intelligibility.

A very basic assumption is that

> the articulation index is based on the concept that any narrow band of speech frequencies of a given intensity carries a contribution to the total index which is independent of the other bands with which it is associated and that the total contribution of all bands is the sum of the contributions of the separate bands. (French and Steinberg, 1947, p. 101)

The observation that led to the theory was made by J.Q. Stewart in 1921 (Fletcher and Galt, 1950, p. 93). Using speech articulation tests with band pass filters, it was noticed that the articulation error scores (c) could be treated as a probability. Thus, the total error was simply a product of the individual errors:

$$c = c_1 \times c_2 \times \ldots c_k \tag{1}$$

Since each individual error term (c_n) can be written in terms of the respective articulation score ($1-s_n$), equation 1 can be rewritten by taking $\log_{10}$ of each side:

$$\log(1-s) = \sum_{n=1}^{k} \log(1-s_n) \tag{2}$$

Equation 2 states that the measure of the total articulation score can be characterized by the sum of the logs of the individual articulation scores. Fletcher (1951) rewrote equation 2 to define the AI as:

$$AI = -\frac{Q}{P}\log(1-s) \tag{3}$$

The negative sign in equation 3 is needed to counteract the logarithm of the error term which is always less than 1. The quantity P is termed the proficiency factor and it reflects the skill of the listener-talker combinations. For many studies involving an articulate speaker and normal listeners the proficiency factor is assumed to be one. Q is a fitting constant that must be adjusted so that AI is between zero and one. Theoretically, a perfect communication system could have a perfect articulation score causing the AI to be infinite. In reality, no system is perfect and the articulation score will never be one.

Although the mathematical treatment of speech intelligibility is insightful, it must be remembered that the AI is an empirical theory. It is helpful then to think about the AI in terms of empirical facts in the way that French and Steinberg presented it in 1947. They assumed that frequency could be divided into 20 bands equally contributing to speech intelligibility. Each band was denoted as A ; thus, each band had a value of .05. The maximum value of the bands was limited by the type of noise or distortion present (W_n) Consequently the total AI was represented by:

$$AI = P\sum_{n=1}^{20} W_n \times A_n \qquad (4)$$

Embedded within equation 4 are all the factors that effect speech intelligibility. The factors explicitly accounted for in the early formulations of the AI were the spectrum of the speakers voices, hearing thresholds of the listeners, interactions between the listener and the speaker, the acoustic and electrical characteristics of anything intervening between the talker and listener and the conditions (noise, speech level etc.) under which the communication was undertaken. The proficiency factor can completely characterize any interactions between the listener and speaker. Both W and A are more complex and need further explanation.

The quantity W for any one band of speech depends upon the auditory sensitivity of the listener and the amount of noise in the environment. It is a measure of the total amount of signal in a frequency band that is available to the listener and thus varies between zero and one. Beranek (1947) and French and Steinberg (1947) assumed that the total dynamic range of speech was 30 dB based upon a study by Dunn and White (1940). Briefly, Dunn and White (1940) made one-eighth second interval root-mean square (rms) pressure measurements of six male and five female speakers. They found that only one percent of all speech peaks exceeded 12 dB above the average long term rms pressure. Speech sounds below 18 dB of the long term rms pressure do not contribute any information (Beranek, 1947, p. 881). Thus, in its simplest form W is simply defined as:

$$W_n = \left(\frac{S_n - M_n}{30}\right) \tag{5}$$

Where S is the effective sensation level of the speech and depends on the listener's acuity and the spectrum of the speech and M is the masking level of the noise. The numerator in equation 5 can be thought of as the speech to noise ratio (SP/N) for the nth band of speech, where 0 dB SP/N is that S/N ratio that produces a zero percent articulation score. The denominator (30 dB) is the assumed total dynamic range of speech, 12 dB above and 18 dB below the long term rms speech level (Beranek, 1947).

French and Steinberg (1947) added what is now called non-linear intensity weighting to the equation. Non-linear weighting assumes that speech sounds of low amplitude will be masked by preceding speech in the same band. Thus, equation 5 becomes:

$$W_n = \left(\frac{S_n - M_n - 6}{30}\right) \tag{6}$$

and assumes that the 30 dB dynamic range of speech does not begin until the speech is 6 dB above the level of Beranek's linear intensity weighting scheme.

French and Steinberg (1947) defined A as equal to .05 since they assumed that frequency could be broken down into 20 equally contributing bands. The width of each band can be determined by articulation testing and also expressed theoretically. If I(f) is the function which expresses at each frequency the importance of that frequency toward speech intelligibility then AI becomes

$$AI = \int_{0}^{\infty} I(f)df \qquad (7)$$

Thus, another way to define the AI is the integral of the frequency importance function over the frequency region of interest. However the masking term W, would still need to be incorporated in a noisy system. Therefore, the AI in its simplest form is a scheme for determining the level of the speech that is both above the threshold of the listener and not masked by the noise. Once the 20 equally contributing frequency bands are determined, it is a simple matter to compute the AI associated with the communication system of interest. The usefulness of this technique was recognized by Kryter (1962a,b) and standardized by the American National Standards Institute (ANSI S3.5-1969).

ANSI S3.5-1969

Speech intelligibility testing is a very expensive and time consuming task. Extensive articulation testing is required to assess a communication system adequately. Kryter (1962a) proposed techniques to use the AI to evaluate most communication systems

without experimentation. The techniques described by Kryter (1962a) were almost entirely adopted in ANSI S3.5-1969. This section will outline some procedures in this standard and introduce alternative methods that have been advocated by other researchers.

Pavlovic and Studebaker (1984) recently examined the approaches for determining the AI reported by Beranek (1947), French and Steinberg (1947) and Kryter (1962a). They found that there were four basic assumptions that differed among the studies. Table 1 documents these parameters along with those adopted in ANSI S3.5-1969.

Table 1: Four parameters that have been questioned in recent research

PARAMETERS

Study	Dynamic Range	Weighting	Peaks	Pauses
Beranek	30	linear	actual	no
French & Steinberg	36	non-linear	both	yes
Kryter	30	linear	both	yes
ANSI S3.5	30	linear	12 dB	yes

The four parameters in Table 1 are dynamic range, weighting, speech peaks and the inclusion or exclusion of speech pauses. The dynamic range is the range of speech information in dB, above and below the rms pressure level of the speech signal. It was taken to be 30 dB (Dunn and White, 1940; Beranek, 1947). French and Steinberg (1947) used a modified dynamic range of 36 dB while all other researchers used 30 dB.

Weighting is the term used to denote the total amount of the AI that is available to the listener. Linear weighting implies that the 30 dB dynamic range is of equal importance. In non-linear weighting, very low levels of speech do not contribute to the AI. French and Steinberg use non-linear weighting whereas other researchers use linear weighting. A more detailed discussion of weighting follows.

A speech peak is defined as 1 percent of the root mean square (rms) levels of the speech signal averaged over one-eighth second intervals that exceed the long term rms level of the speech. There are two ways to incorporate the speech peaks of each band into the AI. One way is to use one averaged number for each band; 12 dB (Dunn and White, 1940); ANSI S3.5-1969 uses this method. The other way is to use the actual speech peaks for each band. Beranek (1947) advocated this method, while both Kryter (1962a) and French and Steinberg (1947) used both techniques.

The parameter of pauses refers to whether the rms averaging of the speech signal included natural pauses. The inclusion of pauses causes the rms pressure level of speech to be about 1 dB below that of speech without pauses (Pavlovic and Studebaker, 1984). Only Beranek excluded the pauses in the analysis.

Pavlovic and Studebaker (1984) recommended one combination of these parameters based upon speech intelligibility tests with a standard nonsense syllable test. These were a 30 dB dynamic range, linear intensity weighting and actual speech peaks. They maintained that the amount of experimental error within the procedure did not allow them to recommend either including or leaving out the natural pauses between words in the test. Unfortunately, even Pavlovic himself later changed position with regards to some of these parameters (Pavlovic, 1987).

As outlined above, there are many ways to implement the AI and questions have been raised about the accuracy of ANSI S3.5-1969 in some situations. A sampling of the

current research reveals many different methods of implementing the ANSI standard. This causes comparisons between research using the AI to be extremely difficult. Still, two of the most prominent questions about the AI, weighting and the importance function can be addressed.

Weighting

The determination of the weighting function is more complex than just choosing either linear or non-linear intensity weighting as Table 1 might imply. French and Steinberg (1947) characterized a quiet listening situation as having an internal noise or X as:

$$X = Q - R \qquad (8)$$

where R is the critical ratio in dB and Q is a pure-tone threshold in quiet. The difference or S, between the internal noise present $(Q-R)$, and the speech spectrum (Y) reaching the ear is defined as:

$$S = Y - Q + R \qquad \text{(eq. 9)}$$

Recall from equations 5 and 6 that S is the effective sensation level of the speech and is used in the calculation of the weighting function W.

ANSI S3.5-1969 uses a different S, or S' to determine when a speech sample is intelligible. Equation 10 defines S' below:

$$S' = Y - Q + C \qquad \text{(eq. 10)}$$

In equation 10, C is the critical band (CB) converted from hertz to decibels. The studies that have used S′ (Kryter, 1962b, ANSI S3.5-1969) tend to overestimate the true signal to threshold level resulting in the articulation scores in noise being greater than those in quiet for the same AI. No such discrepancy exists for studies that have used S (Pavlovic and Studebaker, 1984; Dirks et al., 1986; Pavlovic et al. 1986). As mentioned above, the AI is a measure of speech intelligibility and not of speech detection. Thus, the use of S seems reasonable since it is zero when the energy in the masker and speech sample are equal; whereas, S′ is zero when the speech sample is just detectable.

The Importance Function

The determination of the 20 equally contributing frequency bands to speech can be accomplished by experimentally deriving a frequency importance function as in equation 7. This is done using low and high pass filters and introducing noise to degrade the speech signal. The first frequency importance function was developed by French and Steinberg (1947) by using a combination of female and male speakers articulating consonant-vowel-consonant (CVC) nonsense syllables. Beranek (1947) used the same body of research as French and Steinberg but made use of only the male speakers. Originally it was thought that all types of speech could use the same importance function without significant error (French and Steinberg, 1947, p. 115, Beranek, 1947, p. 885; ANSI S3.5-1969). The increase in articulation scores due to the increase in message redundancy was accounted for by deriving different transfer functions relating AI to articulation scores. However, recent research has documented that there are different importance functions for different

speech material (Black, 1959, Studebaker et al., 1987, Studebaker and Sherbecoe, 1991). This has lead to some questions concerning the practice of using only one frequency importance function for all speech material (Pavlovic, 1987).

Figure 1 illustrates relative importance of frequency bands to intelligibility as reported by Pavlovic (1987) in critical bands for French and Steinberg (1947), Black (1959) and Studebaker et al., (1987) for nonsense syllables, words and continuous discourse, respectively. The most striking difference between nonsense syllables and CD, (the two extremes) is the shift in the peak from 2000-3000 Hz region to that of 400-500 Hz respectively for the nonsense syllables and CD. Also, while the nonsense syllable function is unimodal, both the word and CD functions are bimodal. The introduction of structure as context seems to split the frequency range. A possibly explanation is that fewer consonantal cues need to be utilized from the high frequency regions as more structure is imposed upon the articulation test. Whatever the reason, the use of high or low redundancy importance functions for general articulation tests will add some error across different speech materials. Pavlovic (1987) developed an average frequency importance function that he recommended over any other specific function when performing general articulation tests. Still, the question remains, why not just use an importance function that is best suited for the speech stimulus of interest?

Recent Applications of AI

The AI has been examined by numerous researchers. A brief review of this research exploiting AI follows; in particular, the varying uses of ANSI S3.5-1969.

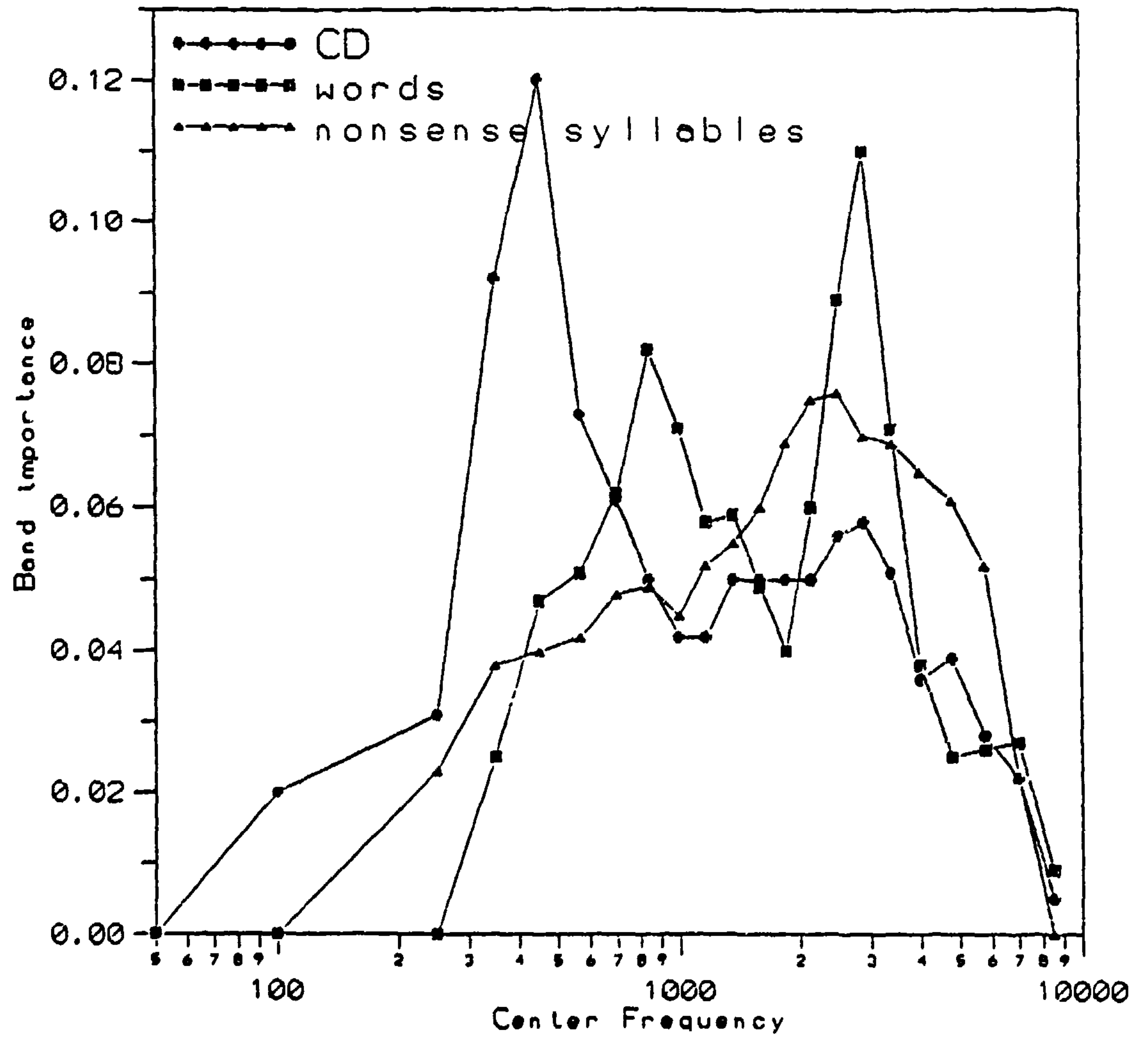


Figure 1: Importance function for nonsense syllables (French and Steinberg, 1947), words (Black, 1959) and CD (Studebaker et al., 1987).

Clinical Applications

The most promising clinical application of AI would be for prescription of hearing aid gain, so that hearing aid fitting would be based on an actual measure of speech intelligibility. Current hearing aid gain prescription methods are based primarily on pure tone thresholds (Berger, 1990). The first step toward hearing aid fitting using the AI concepts would be to modify the AI so that it could be used accurately to predict speech intelligibility for listeners with hearing impairment.

Pavlovic (1984) investigated the use of AI to predict residual auditory function in hearing impaired subjects having a sensironeural (nerve type) hearing loss. He found that hearing loss subjects had disproportionately low speech discrimination compared with an AI prediction of their ability. That is, the greater the subject's hearing loss, the higher the difference between speech discrimination ability and the AI predictions. Pavlovic suggested the introduction of a frequency dependent factor into the AI that would account for the degree of hearing loss.

Kamm et al. (1985) also reached this conclusion. They also attempted to use existing AI methodology to predict speech recognition of hearing impaired subjects. The AI failed to predict the articulation scores for subjects with severe hearing impairments. These results are not surprising since both Pavlovic (1984) and Kamm et al. (1985) used the AI as developed by French and Steinberg (1947), without modification. French and Steinberg mentioned that the validity of AI would be questionable with severely hearing impaired individuals (French and Steinberg, 1947, p. 114).

Pavlovic et al. (1986) investigated the use of a frequency dependent desensitization factor to account for the degree of a subject's impairment. The AI was modified in two

ways. First the weighting factor was changed to account for the fact that subjects with sensironeural hearing impairment have increased critical ratios. Thus, the speech and noise spectrum densities were integrated over larger critical bandwidth than would be done for normal listeners. Second, the contribution of each frequency band to the total AI was the product of the unmodified AI and a speech desensitization factor. The speech desensitization factor was determined experimentally and depended upon the subjects hearing loss. The results of this modified procedure were encouraging. That is, the modified AI predicted intelligibility scores of four moderate to severely impaired subjects within six percent.

Pavlovic (1988) developed a procedure to evaluate different hearing aids and different hearing aid settings on the same hearing aid. He recognized that corrections for the deterioration of suprathreshold speech processing ability is not necessary when results are compared between different hearing aids on the same subject. Thus, the use of an AI predictor, although not necessarily reflecting the actual speech recognition scores, would be useful in comparisons of different hearing aids and different hearing aid settings. He then simplified the frequency importance function, used a dynamic range of 30 dB and a typical speech spectrum corresponding to normal conversational speech. The procedure for calculating the AI becomes a simple matter of determining the amount of speech information that is available by comparing the subject's aided threshold with the speech spectrum. Although the calculated AI did not reflect the acceptability of the aid, it was a promising predictor of speech intelligibility and when used with other indicators can be related to hearing aid satisfaction.

Hearing Protection

It seems natural to consider the use of AI for the prediction of speech intelligibility with the use of hearing protectors. The current method to assess the effectiveness of hearing protectors (ANSI S12.6-1984) provides no consideration to anything but the protectors ability to attenuate sound. The current noise reduction rating (NRR) of hearing protection is a one number figure and is inadequate in providing information to determine which hearing protector will provide the correct balance of attenuation and speech communication (Michael and Bienvenue, 1980). Unfortunately, very little research has been published examining this problem.

Wilde and Humes (1990) investigated the use of AI for making speech intelligibility predictions with normal and hearing impaired listeners fitted with hearing protectors. They used a modified ANSI S3.5-1969 method to calculate the AI. In particular they used French and Steinberg's (1947) importance function instead of Beranek's (1947). Wilde and Humes (1990) found that the AI could be an effective method for predicting the speech intelligibility for normal and mild to severe hearing impaired listeners. Overall, the use of the AI and the known attenuation characteristics of a hearing protector could lead to better hearing protector selection, especially for situations in which speech communication is important.

Williams and Michael (1991) took this one step further. In a study based upon work done by Paul Michael for the steel industry, they attempted to select a hearing protector that would be ideal for a particular kind of noise. This is important since, overprotection can be as ineffective as underprotection in industry. The results are encouraging and the study is still in progress.

Architectural Engineering

It is very common to use AI for determining the placement and design of materials for rooms heavily used for speech communication (Cavanaugh et al., 1962; Herbert, 1978; Pirn, 1971; Warnock, 1978: and Moreland, 1989). There is even an American Society of Testing and Materials (ASTM) standard documenting the test method for evaluating speech privacy in office spaces using AI (ASTM Standard E1130-86).

The most recent use of AI in the design of office spaces by Moreland (1989) typifies much of this research. In particular it demonstrates how, with some imagination, the AI can be used as an effective tool to predict relative speech communication ability. Speech privacy in open offices is a double edged sword. If the office setting is noisy, there will be high speech privacy within each work space; however there also will be low worker satisfaction with the level of noise present. An extremely quiet work space will have poor speech privacy. Thus, some way is needed to specify speech privacy that is easily calculated and that will allow the different parameters effecting work spaces to be evaluated.

The AI provides this measure with only a series of simple measurements. Moreland (1989) placed a loudspeaker in one office space and a microphone in another. A broadband noise was played through the loudspeaker simulating the ideal speech spectrum. The AI was then calculated from the spectrum at the microphone location. In this way, influencing parameters such as the size of office partitions, ceiling material and distance between partitions and the floor could be measured.

Summary

This chapter reviewed literature relating the influence of context to speech intelligibility. Since context is difficult to quantify, it is generally causing researchers to use monosyllabic words as the speech stimuli. The AI concept was reviewed and identified as a method of investigating the relations between spectral context and intelligibility. The weaknesses of the AI were examined and questions concerning its standardization were discussed. Finally, the usefulness of the AI to different applications was reviewed.

Chapter 3

STATEMENT OF THE PROBLEM

The relation between the AI and articulation scores for different types of speech material is illustrated in Figure 2 (reprinted from ANSI S3.5-1969) These data were reported by French and Steinberg (1947) and calculated from data published in Fletcher and Steinberg (1929). Since ANSI S3.5-1969 is used as a method for calculating the AI and a guide for estimating the effect of context, these relations will be examined.

Basic Relation Between Words, Sentences and CD

There are two major reasons to carefully consider the data presented in Figure 2. First, data are based on the calculations performed by French and Steinberg (1947). The Fletcher and Steinberg (1929) data were used as

> subjective tests of the desired character under a variety of conditions where all the required data on the circuits, the speech spectrum, etc., are sufficiently well known to permit computing the articulation index of the received speech. (French and Steinberg, 1947, p. 115)

In other words, an importance function for each type of speech stimulus was not computed. Instead, the importance function for nonsense syllables, derived by French and Steinberg (1947) was used to compute a modified transfer function relating the AI to an articulation score for speech material other than nonsense syllables. The problem with this technique is that the change in message redundancy or change in context will greatly influence the importance function (Pavlovic, 1984).

The weakness of French and Steinberg's approach is further illustrated in Figure 1. The addition of context to the speech material shifts the peak area of the importance function. It also changes the shape of the importance function from unimodal to bimodal

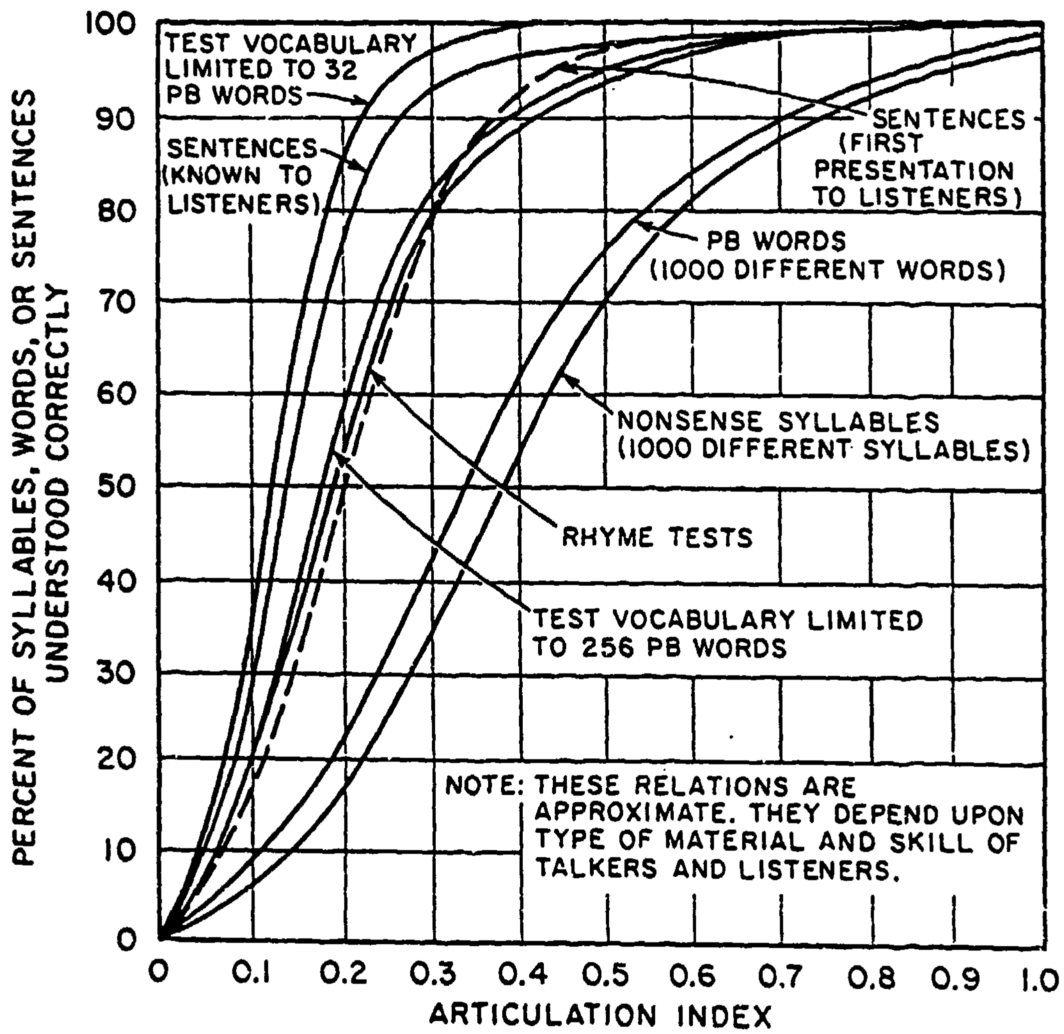


Figure 2: Relation between AI and articulation score for different types of speech material, figure 15 of ANSI S3.5-1969. Reprinted by permission of the Acoustical Society of America, New York, New York.

for sentences and CD material. Thus, calculation of AI versus articulation score (transfer function) values, which depend upon a frequency importance function that is not specific to the speech material type, would inaccurately reflect the articulation score (Boothroyd, 1978; Miller and Nicely, 1955).

The second reason to reassess the results illustrated in Figure 2 is that the results were obtained without the use of recording technology. Thus, each utterance was a unique event and invariably each subject received a different test. Kruel et al. (1969) reported that the same list of words spoken by two different speakers produces two different tests. Thus, the lack of control of speaker variability could add considerable error into an experiment.

Surprisingly, only the French and Steinberg (1947) research documents the relation between AI and articulation scores for different types of speech material. Other studies have documented the AI for one type of speech material, such as words (Black, 1959), distinctive features (Duggirala et al., 1988) and CD (Studebaker et al., 1987). It is unfortunate that methodological differences between these studies do not allow for any specific relations to be derived between the respective types of speech.

Experimental Objectives

The purpose of this research was to determine some basic relations between words, sentences and CD. The articulation index was used so that the frequency importance and transfer functions for words, sentences and continuous discourse could be defined. Another goal of this research was to define the range of speech intelligibility from words to sentences to continuous discourse, for one speaker, using the same equipment and the same subjects under very controlled conditions. The experimental objectives were to:

1. Derive transfer functions relating the AI to articulation scores for words, sentences and CD;

2. Compare transfer functions for a word identification task to a sentence and CD estimation task; and,

3. Develop frequency importance functions for words, sentences and CD.

Chapter 4

METHODS

Subjects

Thirty-two subjects between 18 and 35 years were in the initial subject pool. Each subject had normal air conduction hearing thresholds ($\leq$ 15 dB HL re ANSI S3.6-1989) from .125 and 8.0 kHz in octave intervals in each ear and the difference between their thresholds did not differ at any freouency by 10 dB between ears. Each subject was a native speaker of American English.

However, eight subjects were eliminated from the study for reasons explained in the reliability section. Consequently, 24 subjects (14 males, 10 females) having a mean age of 26.9 years (std. dev. of 4.4) participated. The experiment was conducted individually in three 75 minute sessions, with no more than two days separating each succesive session. The subjects were paid for their participation.

Speech Stimuli

The test stimuli were monosyllabic words, meaningful sentences and continuous discourse (CD). Specifically, 616 monosyllabic words taken from the PB-50 word lists (ANSI S3.1, 1991) were used. The meaningful sentences were 200 sentences from the revised SPIN test (Bilger, 1985). The extensive testing in the development of the SPIN sentences (Kalikow, 1977, Owen, 1981, Morgan et al., 1981 and Bilger, 1985) was the deciding factor for its use. There were 44 CD passages. Each passage was 20 to 25 seconds in length. All passages were taken from children's encyclopedias and conformed to

a seventh-grade reading level (Fry, 1968). The subject matter of the CD passages concerned common objects such as plants, animals, household objects, food, etc. Each CD passage also contained the title of the passage in the first sentence.

Instrumentation

Live-voice recordings of the speech stimuli were made in an audiometric test booth having ambient noise levels suitable for ears open testing (ANSI S3.1-1977). All speech stimuli were spoken by one male articulate speaker having a general American dialect. The recordings were made using a broadcast quality microphone (Electrovoice PL-10) connected to a digital audio tape (DAT) recorder (Sony TCD-10). The speaker was approximately 0.5 meter from the microphone and used normal vocal effort (65 dB SPL) to produce a target of 0 dB VU while watching a large VU meter. The master recordings were then transferred onto optical storage via a digital signal processing (DSP) board (Ariel DSP-16) having a sampling rate of 18116 Hz. The whole process was controlled by a personal computer (AT&T 6300). Before digitizing the speech was filtered by an 8 KHz low pass 8-pole Butterworth anti-aliasing filter. Playback used the DSP board through the same 8 kHz low pass filter.

Variable filtering of the speech stimuli was accomplished by the DSP board as follows. The algorithm reported by Defatta et al. (1988) was programmed in Turbo Pascal to obtain filter coefficients for finite impulse response (FIR) high pass (HP) and low pass (LP) filters. All filter coefficients implemented filters with a transition band of 60 dB per octave and a maximum ripple of 0.1 dB. The filter coefficients were converted to hexadecimal values and utilized in a program that was written in Texas Instruments TMS3020 assembly language (the Ariel DSP-16 uses the TMS3020 DSP integrated circuit). The DSP

board was controlled by software written in Turbo Pascal in the MS-DOS environment. The implementation of the filters in assembler code was necessary to allow the filtering of the speech stimuli to be carried out without excessive delays during a subject's test session. The speech signal was then mixed with a masker and amplified. The programs to control the DSP board and filter the speech are included in Appendix D.

The masker was white noise shaped to the one-third octave band speech peaks of the speaker. The speaker's one-third octave rms level were determined with a real time analyzer (Bruel and Kjaer 3347). The level of the speech peaks above the rms were estimated as per Dunn and White (1940) and added to the rms levels to produce estimated speech peaks for each one-third octave band. The masker was generated by a one-third octave noise generator and recorded onto a DAT tape. The desired signal to noise (S/N) ratio was obtained by varying the level of the shaped white noise.

The speech stimuli and the masker were then mixed using a custom built summing amp and delivered to both ears via TDH-39 earphones equipped with MX-41/AR supra-aural cushions. The frequency response of the earphones was measured before and after the study and both curves were within 1.5 dB. A calibration check of the earphones was performed each day that tests were run. This was done using a 1 kHz test tone, played from the DSP board, corresponding to the speakers target 0 dB VU on the same VU meter used by the speaker during live-voice recording. All experimentation was conducted in a sound treated room conforming to ANSI S3.1-1977 permissible ambient noise levels for audiometric testing under phones.

Data Collection

Words, Sentences and Continuous Discourse

Each of the subjects' sessions were controlled by a personal computer (PC). A program was written in Turbo Pascal (see Appendix D) that used previously digitized words, sentences and CDs. The program loaded each speech type individually, filtering the data according to the subject's randomized data file. These data files randomized the stimulus, filter and S/N ratio order. The filtering took place while the digitized speech sample files were transfered from DOS to the DSP boards memory buffer. The program output the randomized noise setting to the screen. The experimenter manually set the S/N ratio using an attenuator box (Hewlett Packard 350D).

PB Words

The PB words were presented in random order with the unfiltered carrier phrase "Will you write" preceding each word. All 616 words were presented to each subject in sets of 14, one test condition at a time. The subjects were required to write down each word on a response sheet.

Sentences

The 176 sentences were also presented in random order to each subject in blocks of four. The subject's task was to estimate the percentage of words correctly understood between 0 and 100 percent as well as to write down the key word in each sentence. (Speaks et al., 1972). All estimates were made in 7.5% increments.

An estimation task was chosen in place of a key-word in sentence identification task due to the small number of sentences available for testing. It was assumed that a subject could optimally make estimates of the percentage of words understood correctly in the CD task to within ± 7.5%. Consequently, at least 7 binary choices (eg., each SPIN sentences) were necessary to make the binary task comparable to the sensitivity of the CD task. A doubling of that number to 14 words or sentences would assure the sensitivity of the binary task would be at least that of the estimation task. There were 616 PB words available, but only 200 sentences available for testing. Thus an estimation procedure of percent correct was assumed to be a more accurate prediction of sentence intelligibility. This assumption is supported by Cox, Alexander and Rivera (1991).

CD Passages

The subjects were instructed to listen to a CD passage of 20-25 second length and then to estimate in increments of 7.5, the percentage of words that they understood between 0 and 100. The assertion that this method produces a valid estimate of the intelligibility of CD is supported by Studebaker et al. (1982) and Cox and McDaniel (1984).

Test Procedures

Each subject received a total of 44 test conditions for each type of speech (11 filter conditions x 4 S/N ratio conditions). The filter conditions were low pass (LP) filters having cutoffs at 400, 650, 920, 1400 and 4500 Hz and high pass (HP) filters having cutoffs at 920, 1400, 2200, 3000, and 5500 Hz. The four S/N ratios used in this study were determined as a result of a pilot experiment using eight subjects. The subjects in the pilot

study performed the same tasks as the 24 subjects used in this study, except that they only made estimates of unfiltered speech in noise. The S/N ratio levels for the words were +9, +3, -3 and -9, for the sentences were 0, -4, -8 and -12 and for the CD were -3, -6, -9 and -12. The S/N ratios were measured as follows. A 1 kHz tone was digitized by the DSP board that corresponded to the talker's target 0 VU. This tone was used to set the level of the speech. The noise was passed through an attenuator box and output levels which corresponded to the desired S/N ratios were determined manually using a sound level meter with a type NBS-9A type coupler (General Radio, 1565-z).

There were 14 binary decisions from each subject at each test condition for the words. Since there were 44 CD and 44 sets of four sentences, each subject contributed one intelligibility estimate at each test condition for CD and sentences. Therefore, there were 24 estimates for sentences and CD and 24 scores based upon 14 words for the word recognition task.

The order of presentation of types of speech was counterbalanced using the basic sequence of words-sentences-CD for the first 12 subjects and sentences-words-CD for the last 12 subjects. The 44 test conditions were presented randomly for each speech type to each subject over three sessions. The speech was presented at an overall level of 73 dB SPL, initially calibrated in an NBS-9A coupler, in the unfiltered condition. This level represents a normal conversational level in the ear canal (Pavlovic, 1984). The noise was then added and attenuated to control the S/N ratio.

Subjects' Task

Each subject received written instructions (Appendix F) at the beginning of the experiment. Subjects' questions were encouraged and were answered by the experimenter by rereading the appropriate section of the written instructions. If the subject was still

Table 2: Data generated from each subject

Type	Eleven Filters
CD	1 estimate at each of four S/N ratios
Sentences	1 estimate at each of four S/N ratios & % identification based upon 4 key words
Words	% score based upon 14 words

unsure of the task, the written instructions were restated until the subject understood the task.

Prior to making any CD and sentence estimates, each subject practiced making intelligibility judgments. Two passages, two sets of four sentences and two blocks of fourteen words were used with various filter and noise conditions that represented the range of difficulty present in the test sessions. The practice stimuli were not repeated in the test sessions. At the completion of the practice session the subject was allowed to ask questions regarding the task required of them. At this point the experiment began.

Reliability

During each test session the subject was exposed to a single noise and filtered condition. The noise and filtering condition was chosen randomly, the only restriction being that the mean of the condition be between 30 and 70%. Conditions between 30 and 70% were chosen based upon data from the first 10 subjects. This restriction ensured that the

variability of the reliability condition was high. Also, one condition within a session was presented twice. This was done to assess the reliability of each subject's estimates (within session) and to evaluate any criterion shift of a subject (across sessions). One subject exhibited an extraordinarily large criterion shift of 75% and was eliminated. Four subjects had a large within session test score difference (>22.5%) and were dismissed. Finally, three subjects were dismissed since their reliability estimates were made at conditions that were not between the required mean score of 30 to 70%.

Chapter 5

RESULTS

Word Recognition

Each subject's word recognition score (WRS) was obtained by determining the percentage of 14 PB words, presented from a pool of 616 PB words, that they correctly recognized in each experimental condition. Each subject's WRS for each experimental condition (raw data) is shown in Appendix A. Table 3 shows the mean and median WRSs, standard deviations and range scores for each signal to noise (S/N) ratio and filtering condition. WRSs were not obtained for the low pass (LP) cutoff frequency at LP400, -9 and -3 dB S/N ratios and LP650, -9 dB S/N ratio because of their severe difficulty. Inspection of Table 3 reveals that as a general rule, the mean WRSs increased as the S/N ratio became less severe (-9 to +9 dB) for each filtering condition. Further, the mean WRSs for each S/N ratio increased as the LP filter cutoff frequency increased and as the high pass (HP) filter cutoff frequency decreased. Generally, the highest WRSs were obtained in the all-pass filter condition while the lowest WRSs were obtained in the narrowest LP and HP filter condition regardless of S/N ratio. Close inspection of Table 3 reveals that the median WRSs were very similar to the mean WRSs, for each S/N ratio and filtering condition. This finding would tend to indicate that the WRSs were normally distributed around the mean for each S/N ratio and filtering condition. For each S/N ratio, the standard deviation and range scores tended to increase as the LP filter cutoff frequency increased and as the HP filter cutoff frequency decreased. In part, this finding occurred because the WRSs obtained in the narrower LP and HP pass filter conditions were close to 0%. Consequently, the variability of the WRSs were limited. However, as

Table 3: Mean and median WRSs and standard deviations and range scores for each S/N ratio and filtering condition. Tabled values in percentage, N=24.

S/N	Filter	Mean	Median	St. dev.	Range
-9 dB	100-400	-	-	-	-
	100-650	-	-	-	-
	100-920	0.0	0.0	0.0	-
	100-1400	0.3	0.0	1.4	7.1
	100-4500	6.2	7.1	8.1	35.7
	all pass	11.9	14.3	8.6	28.6
	920-8000	16.1	14.3	12.4	42.9
	1400-8000	6.2	7.1	6.6	21.4
	2200-8000	4.5	0.0	5.8	21.4
	3000-8000	1.2	0.0	2.6	7.1
	5500-8000	0.6	0.0	2.0	7.1
-3 dB	100-400	-	-	-	-
	100-650	0.6	0.0	2.0	7.1
	100-920	2.4	0.0	3.3	7.1
	100-1400	7.4	7.1	7.8	28.6
	100-4500	46.4	42.9	15.2	64.3
	all pass	50.0	50.0	14.9	64.3
	920-8000	40.3	42.9	13.1	57.0
	1400-8000	40.2	39.3	15.0	50.0
	2200-8000	17.0	14.3	7.4	28.6
	3000-8000	10.7	10.7	6.8	21.4
	5500-8000	1.8	0.0	3.7	7.1
+3 dB	100-400	0.3	0.0	1.4	7.1
	100-650	2.7	0.0	5.1	21.4
	100-920	9.8	7.1	7.9	28.6
	100-1400	29.7	28.6	11.1	42.9
	100-4500	73.6	78.5	16.8	57.1
	all pass	83.3	85.7	7.9	28.6
	920-8000	66.7	71.4	10.9	50.0
	1400-8000	61.0	60.7	11.1	42.9
	2200-8000	33.1	35.7	15.8	57.5
	3000-8000	14.9	14.3	10.7	35.7
	5500-8000	4.1	7.1	4.1	14.3
+9 dB	100-400	1.5	0.0	2.9	7.1
	100-650	9.2	7.1	7.6	28.4
	100-920	24.5	21.4	12.1	50.0
	100-1400	54.8	57.1	15.3	50.0
	100-4500	89.0	89.3	8.3	28.6
	all pass	90.5	92.9	9.4	35.9
	920-8000	81.0	78.6	10.9	50.0
	1400-8000	79.5	85.7	12.9	50.0
	2200-8000	49.2	50.0	13.3	42.7
	3000-8000	25.0	24.9	12.0	42.7
	5500-8000	3.0	0.0	5.8	21.4

more frequency information became available to the subjects for each S/N ratio, the mean WRSs increased and the variability also increased.

Sentence Intelligibility

Each subject's sentence intelligibility score (SIS) was obtained using four sentences, from a pool of 176 sentences, in each experimental condition. Each subject listened to the four sentences and then estimated the percentage of words they correctly understood in 7.5% increments between 0 and 100%. Each subject's SIS for each experimental condition is shown in Appendix B. Table 4 shows a summary of the subject's SISs presenting the mean and median SISs, standard deviations, and range scores for each S/N ratio and filtering condition. Three SISs were not obtained (LP400 at S/N=-12 and -8 dB and LP650 at S/N=-12 dB) because these conditions were too severe. Inspection of Table 4 reveals similar observations as made for the WRSs. That is, as a general rule, :1) the mean SISs increased as the S/N ratio became less severe (-12 to 0 dB) for each filtering condition, 2) the mean SISs for each S/N ratio increased as the LP filter cutoff frequency increased and as the HP filter cutoff frequency decreased, 3) the highest SISs were obtained in the all-pass filter condition while the lowest SISs were obtained in the narrowest LP and HP pass filter condtion regardless of S/N ratio, 4) the median SISs were very similar to the mean SISs, for each S/N ratio and filtering condition, and 5) for each S/N ratio, the standard deviation and range scores tended to increase as the LP filter cutoff frequency increased and as the HP filter cutoff frequency decreased.

Table 4: Mean and median SISs and standard deviations and range scores for each S/N ratio and filtering condition. Tabled values in percentage, N=24.

S/N	Filter	Mean	Median	St. dev.	Range
-12 dB	100-400	-	-	-	-
	100-650	-	-	-	-
	100-920	0.0	0.0	0.0	-
	100-1400	5.0	0.0	6.0	7.1
	100-4500	16.6	11.3	15.0	67.5
	all pass	26.3	26.3	18.8	60.0
	920-8000	26.3	22.5	16.6	52.5
	1400-8000	18.1	11.3	19.2	67.5
	2200-8000	6.6	7.5	5.4	15.0
	3000-8000	4.1	0.0	7.5	22.5
	5500-8000	2.2	0.0	5.1	22.5
-8 dB	100-400	-	-	-	-
	100-650	0.0	0.0	0.0	-
	100-920	0.9	0.0	2.5	7.5
	100-1400	13.4	7.5	12.8	45.0
	100-4500	48.4	45.0	24.6	90.0
	all pass	62.8	67.5	23.3	90.0
	920-8000	49.7	52.5	22.6	97.5
	1400-8000	48.4	52.5	24.1	90.0
	2200-8000	20.0	15.0	17.5	67.5
	3000-8000	8.1	7.5	8.6	30.0
	5500-8000	5.0	0.0	7.7	30.0
-4 dB	100-400	0.0	0.0	0.0	-
	100-650	4.1	0.0	8.4	30.0
	100-920	10.3	3.8	14.0	35.0
	100-1400	50.3	48.8	13.4	60.0
	100-4500	91.4	97.5	11.9	47.5
	all pass	95.8	100.0	6.2	17.5
	920-8000	91.1	93.8	8.3	25.0
	1400-8000	86.8	90.0	11.7	40.0
	2200-8000	45.6	45.0	21.5	82.5
	3000-8000	25.3	22.5	19.3	60.0
	5500-8000	9.7	7.5	13.3	60.0
0 dB	100-400	3.4	0.0	5.7	22.5
	100-650	8.4	7.5	7.9	37.5
	100-920	33.4	30.0	20.8	75.0
	100-1400	80.9	82.5	13.5	55.0
	100-4500	99.2	100.0	2.1	10.0
	all pass	99.7	100.0	0.8	2.5
	920-8000	98.3	100.0	3.5	17.5
	1400-8000	95.8	100.0	7.6	25.0
	2200-8000	71.3	78.8	21.1	75.0
	3000-8000	40.3	37.5	20.4	75.0
	5500-8000	14.0	7.5	14.7	52.5

Connected Discourse Intelligibility

Each subject's connected discourse intelligibility score (CDIS) was obtained using one connected discourse (CD) passage, from a pool of 44 passages, in each experimental condition. For each CD passage, each subject estimated the percentage of words they understood in 7.5% increments from 0 to 100%, Each subject's CDIS is shown in Appendix C. Table 5 shows a summary of the subject's mean and median CDISs, standard deviations, and range scores for each S/N ratio and filtering condition. Three of the experimental conditions (LP400 at S/N=-12 and -9 dB; LP650 at S/N=-12 dB) were not presented due to their severe difficulty. Inspection of Table 5 reveals similar observations made for WRSs and SISs. Briefly, the mean CDISs increased as the S/N ratio became less severe (-12 to -3 dB) for each filtering condition and, the mean CDISs for each S/N ratio increased as the LP filter cutoff frequency increased and as the HP filter cutoff frequency decreased. Generally, the highest CDISs were obtained in the all-pass filter condition while the lowest CDISs were obtained in the narrowest LP and HP filter condtion regardless of S/N ratio. The median CDISs were very similar to the mean CDISs for each S/N ratio and filtering condition and for each S/N ratio, the standard deviation and range scores tended to increase as the LP filter cutoff frequency increased and as the HP filter cutoff frequency decreased. finally, as more frequency information became available for each S/N ratio, the mean CDISs and the variability increased.

Table 5: Mean and median CDISs and standard deviations and range scores for each S/N ratio and filtering condition. Tabled values in percentage, N=24.

S/N	Filter	Mean	Median	St. dev.	Range
-12 dB	100-400	-	-	-	-
	100-650	-	-	-	-
	100-920	0.0	0.0	0.0	-
	100-1400	1.6	0.0	3.7	15.0
	100-4500	12.2	7.5	13.1	52.5
	all pass	21.3	15.0	17.0	75.0
	920-8000	15.6	7.5	13.5	52.5
	1400-8000	13.4	7.5	15.0	60.0
	2200-8000	7.5	7.5	7.2	22.5
	3000-8000	1.6	0.0	3.7	15.0
	5500-8000	0.9	0.0	2.5	7.5
-9 dB	100-400	-	-	-	-
	100-650	0.0	0.0	0.0	-
	100-920	2.2	0.0	6.3	30.0
	100-1400	0.9	0.0	3.3	15.0
	100-4500	26.9	26.3	21.3	82.5
	all pass	43.8	45.0	21.3	70.0
	920-8000	34.4	30.0	21.1	82.5
	1400-8000	22.5	15.0	22.4	82.5
	2200-8000	10.6	7.5	11.0	45.0
	3000-8000	3.1	0.0	5.7	22.5
	5500-8000	0.0	0.0	0.0	-
-6 dB	100-400	0.0	0.0	0.0	-
	100-650	0.0	0.0	0.0	-
	100-920	8.1	7.5	9.9	45.0
	100-1400	11.9	7.5	13.5	52.5
	100-4500	76.4	82.5	20.0	55.0
	all pass	86.5	90.0	12.5	47.5
	920-8000	92.2	97.5	10.3	47.5
	1400-8000	60.9	63.8	23.0	75.0
	2200-8000	39.7	37.5	21.4	75.0
	3000-8000	9.4	7.5	10.0	30.0
	5500-8000	2.5	0.0	3.5	7.5
-3 dB	100-400	0.3	0.0	1.5	7.5
	100-650	3.4	0.0	3.7	7.5
	100-920	18.0	11.3	16.3	60.0
	100-1400	46.1	52.5	26.8	82.5
	100-4500	95.1	100.0	9.6	40.0
	all pass	96.5	97.5	5.7	25.0
	920-8000	92.6	97.5	11.2	40.0
	1400-8000	89.2	90.0	7.7	32.5
	2200-8000	55.6	56.3	20.4	90.0
	3000-8000	20.0	18.8	16.4	67.5
	5500-8000	6.9	7.5	8.6	37.5

Data Transformation

Statistically, it is well known that when WRSs, SISs, and CDISs are expressed in proportionate or percentage scores the means and variances are correlated (Neter, Wasserman and Kutner, 1985). As such, they are not suited to descriptive or inferential statistics because the data are for the most part non-linear and non-additive in a probabilistic sense. Consequently, each subject's WRS, SIS and CDIS (raw data) were transformed using an arcsine transformation (Studebaker, 1985). An example of this procedure is included in Appendix E. The transformed scores then were used to compute mean WRSs, SISs and CDISs for each experimental condition. Then the mean WRSs, SISs and CDISs were transformed back into percentage scores.

Tables 6, 7 and 8 show and figures 3, 4 and 5 illustrate the subject's mean WRS, SIS and CDIS respectively for each filtering and S/N ratio condition in percent following the inverse arcsine transformation. The smoothed lines connecting the mean WRSs, SISs, and CDISs in figures 3, 4 and 5 respectively were generated using a cubic spline procedure (Grapher, 1988). In each figure, the smoothed line between each data point was adjusted so the connecting line was not higher or lower than the actual data point. Overall, this had the effect of smoothing the WRSs, SISs and CDISs curves for each S/N ratio condition when plotted as a function of filter cut-off frequency.

Inspection of Table 6 and Figure 3 reveals that the transformed mean WRSs increased for each filtering condition as the S/N ratio went from -9 to +9 dB. Stated another way, as the S/N ratio became more favorable, the WRSs for each filtering condition increased. Generally, in each S/N ratio, the highest WRSs occurred for the widest LP and HP filter cutoff frequencies and in the all-pass conditions. Further, for each S/N

Table 6: Mean WRSs for each filtering and S/N ratio condition. Tabled values are in percentage following an arcsine transformation.

	SIGNAL TO NOISE RATIO			
FILTER	-9	-3	+3	+9
100-400	-	-	0.0	0.3
100-650	-	0.1	0.7	7.0
100-920	0.0	0.8	7.4	22.7
100-1400	0.0	4.5	29.1	55.1
100-4500	3.2	46.3	76.4	91.5
all-pass	9.2	49.9	84.6	93.9
920-8000	12.8	39.8	67.1	82.4
1400-8000	3.6	39.6	61.3	81.4
2200-8000	2.0	15.8	31.9	49.2
3000-8000	0.2	8.7	11.9	23.9
5500-8000	0.1	0.4	2.3	0.7

Table 7: Mean SISs for each filtering and S/N ratio condition. Tabled values are in percentage following an arcsine transformation.

	SIGNAL TO NOISE RATIO			
FILTER	-12	-8	-4	0
100-400	-	-	0.0	1.1
100-650	-	0.0	1.0	6.1
100-920	0.0	0.1	5.0	31.1
100-1400	2.3	9.2	50.4	84.1
100-4500	13.5	48.4	94.9	99.8
all-pass	22.8	64.0	98.4	100.0
920-8000	23.3	49.1	93.8	99.8
1400-8000	13.4	46.3	89.5	98.5
2200-8000	4.3	17.5	44.4	73.4
3000-8000	1.0	4.9	20.6	38.3
5500-8000	0.4	2.0	5.4	9.5

Table 8: Mean SISs for each filtering and S/N ratio condition. Tabled values are in percentage following an arcsine transformation.

	SIGNAL TO NOISE RATIO			
FILTER	-12	-8	-4	0
100-400	-	-	0.0	0.0
100-650	-	0.0	0.0	1.6
100-920	0.0	0.3	4.8	14.5
100-1400	0.3	0.1	7.8	47.6
100-4500	8.5	22.3	81.4	98.2
all-pass	18.4	42.7	91.9	98.4
920-8000	12.6	32.1	95.1	96.5
1400-8000	9.0	17.7	63.2	90.6
2200-8000	4.6	6.8	38.6	56.3
3000-8000	0.3	0.9	5.4	16.6
5500-8000	0.1	0.0	0.9	3.6

ratio condition, the WRSs increased from the narrowest to the widest LP and HP filter cutoff frequencies. This finding can be related to the amount of frequency information available to the subjects. More specifically, using the subject's mean WRSs as a function of filter cutoff frequency, a crossover frequency can be estimated. That is, a crossover frequency which divides the available frequency information into two equal parts can be estimated for each S/N ratio. Close inspection of Figure 3 reveals that for each S/N ratio the subjects mean WRSs intersected for the LP and HP cutoff filter frequency. This intersection can be called the crossover frequency. The crossover frequency was estimated to be 2760, 2360, 2144, and 2040 Hz for the -9, -3, +3 and +3 dB S/N ratios respectively. Consequently, the crossover frequency decreased as the S/N ratio became more favorable (-9 dB to +9 dB).

Inspection of Table 7 and Figure 4 reveals that the transformed mean SISs increased for each filtering condition as the S/N ratio went from -12 to 0 dB. Stated another way, as

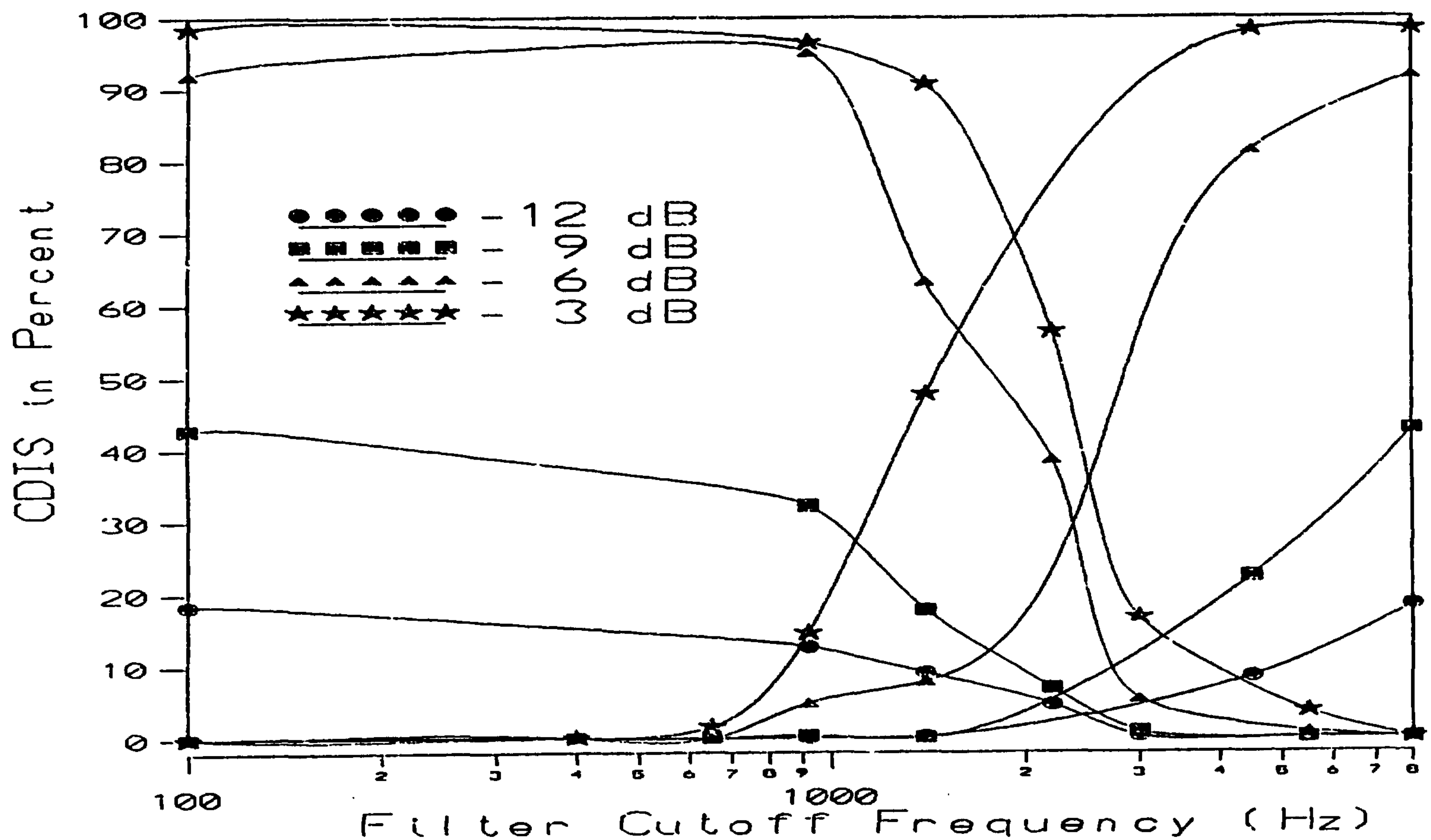


Figure 5: Mean CDISs for each S/N ratio plotted in reference to the filter cutoff frequency.

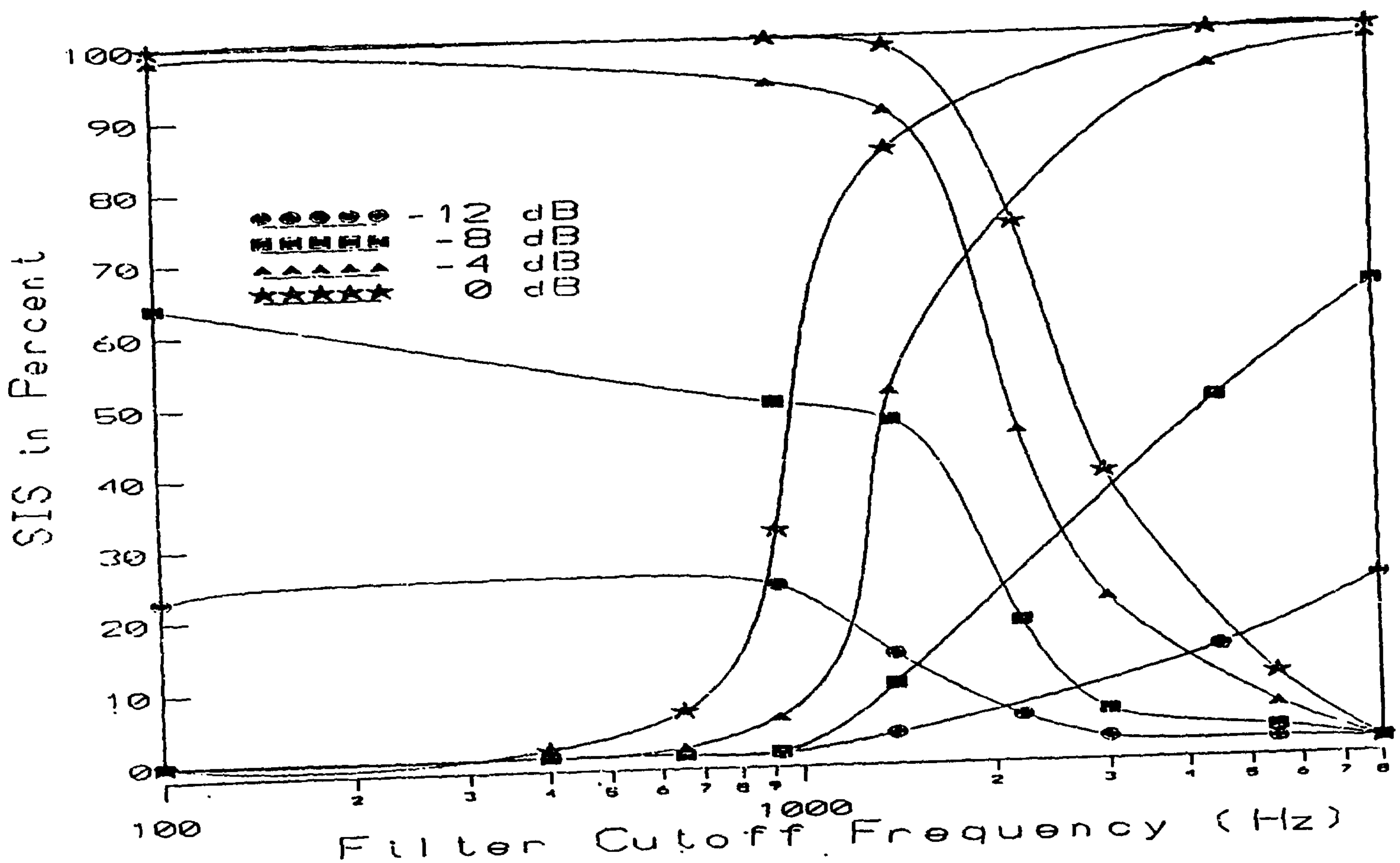


Figure 4: Mean SISs for each S/N ratio plotted in reference to the filter cutoff frequency.

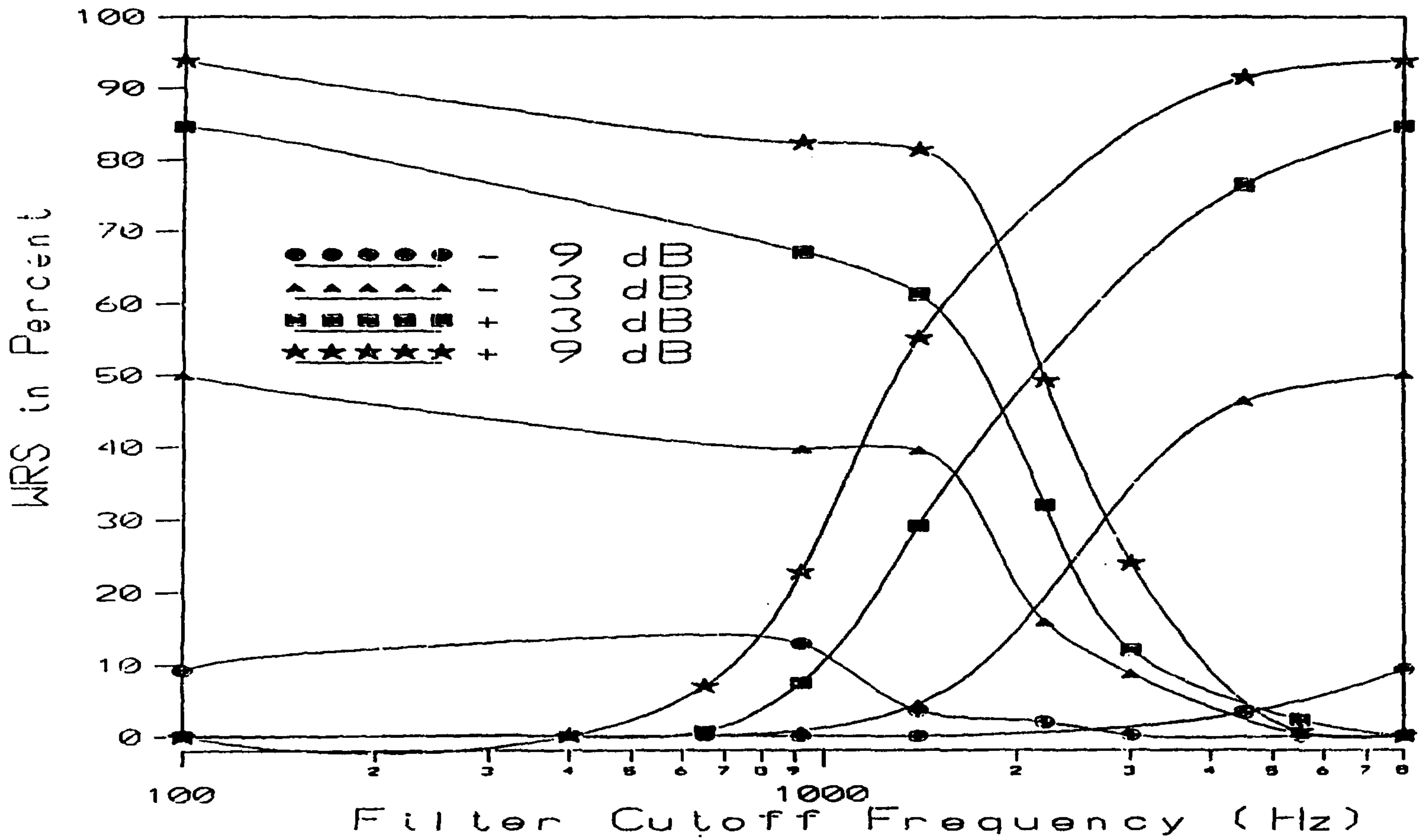


Figure 3: Mean WRSs for each S/N ratio plotted in reference to the filter cutoff frequency.

the S/N ratio became more favorable, the SISs for each filtering condition increased. Generally, in each S/N ratio, the highest SISs occurred for the widest LP and HP filter cutoff frequencies and in the all-pass conditions. Further, for each S/N ratio condition, the SISs increased from the narrowest to the widest LP and HP filter cutoff frequencies. The crossover frequency was estimated to be 2318, 2313, 2054, and 1890 Hz for the -12, -8, -4, and 0 dB S/N ratios respectively.

Inspection of Table 8 and Figure 5 reveals the transformed mean CDISs increased for each filtering condition as the S/N ratio went from -12 to -3 dB. Generally, in each S/N ratio, the highest CDISs occurred for the widest LP and HP filter cutoff frequencies and in the all-pass conditions. Further, for each S/N ratio condition, the CDISs increased from the narrowest to the widest LP and HP filter cutoff frequencies. The crossover frequency was estimated to be 2633, 2501, 2501, and 2219 Hz for the -12, -9, -6, and -3 dB S/N ratios respectively.

A comparison of the mean WRSs, SISs and CDISs shown in Tables 3, 4 and 5 respectively with the mean transformed WRSs, SISs and CDISs shown in Tables 6, 7 and 8 reveals that the arcsine transformation did not change the relations among the WRSs, SISs and CDISs previously discussed. In general, as the listening condition became more difficult, either due to the frequency band limiting or the addition of noise, the WRSs, SISs and CDISs became lower. Also, because the crossover frequencies became higher in frequency as the S/N ratio became more favorable (more positive) for all of the types of speech, the crossover frequency shifts due to noise were not dependent upon speech type.

Derivation of the Relative Transfer Function

Recall, the articulation index (AI) can be used as a measure of the amount of acoustic information that is available to a listener. Further, a relative or absolute transfer function can be calculated to equate the amount of acoustic information or AI which must be present for a listener to achieve a certain WRS, SIS or CDIS. A relative transfer function assumes that the maximum AI or maximum amount of acoustic information available to a listener is equal to one. An absolute transfer function will have the same slope and shape as a relative transfer function. However, an absolute transfer function will be shifted to account for the fact that, even at the S/N ratio that produced the highest WRSs, SISs and CDISs, some of the acoustic information was not available to the subjects.

Figure 7 illustrates the relative transfer functions for the subject's WRSs, SISs and CDISs. The relative transfer functions were developed in the same way as originally proposed by French and Steinberg (1947). In order to do this it was necessary to plot the subjects mean transformed WRSs, SISs and CDISs for each S/N ratio as shown in Figure 6. Figure 6 reveals that as the S/N ratio becomes more favorable (-17 to 10 dB), WRSs, SISs and CDISs increased. The functions for the SISs and CDISs were very similar and increased at larger percent per dB than the WRSs. Close inspection of Figure 6 reveals that each function consists of eight WRSs, SISs and CDISs. Four of these scores were obtained from the 24 subjects used in the experiment. The other four scores were obtained either from a pilot study (N=8; performed to set the experimental S/N ratios), or from subsets of subjects (N=8) who participated in the present study. The graphical methods used by French and Steinberg (1947) involve two basic techniques from which a relative transfer function can be derived. For the purpose of this study, these techniques are known

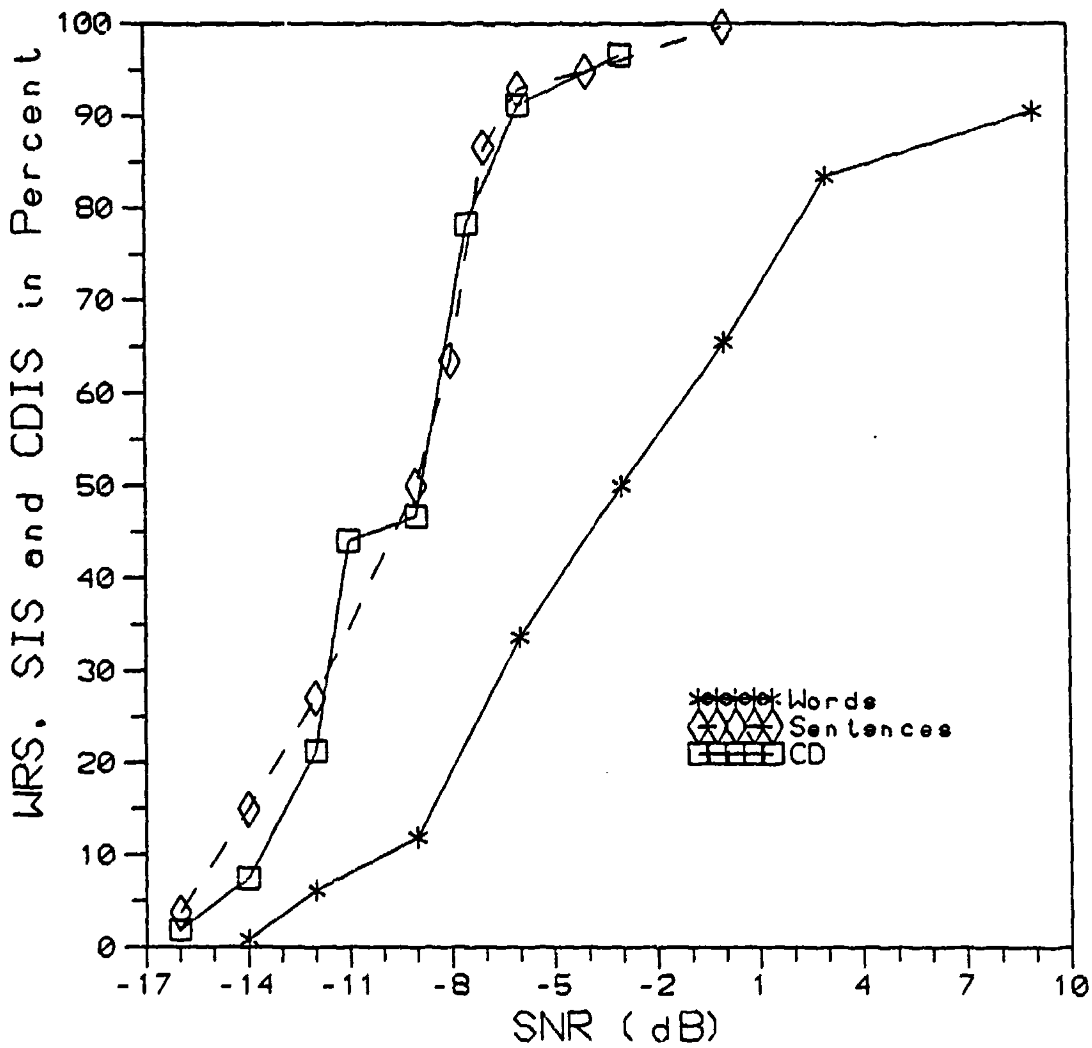
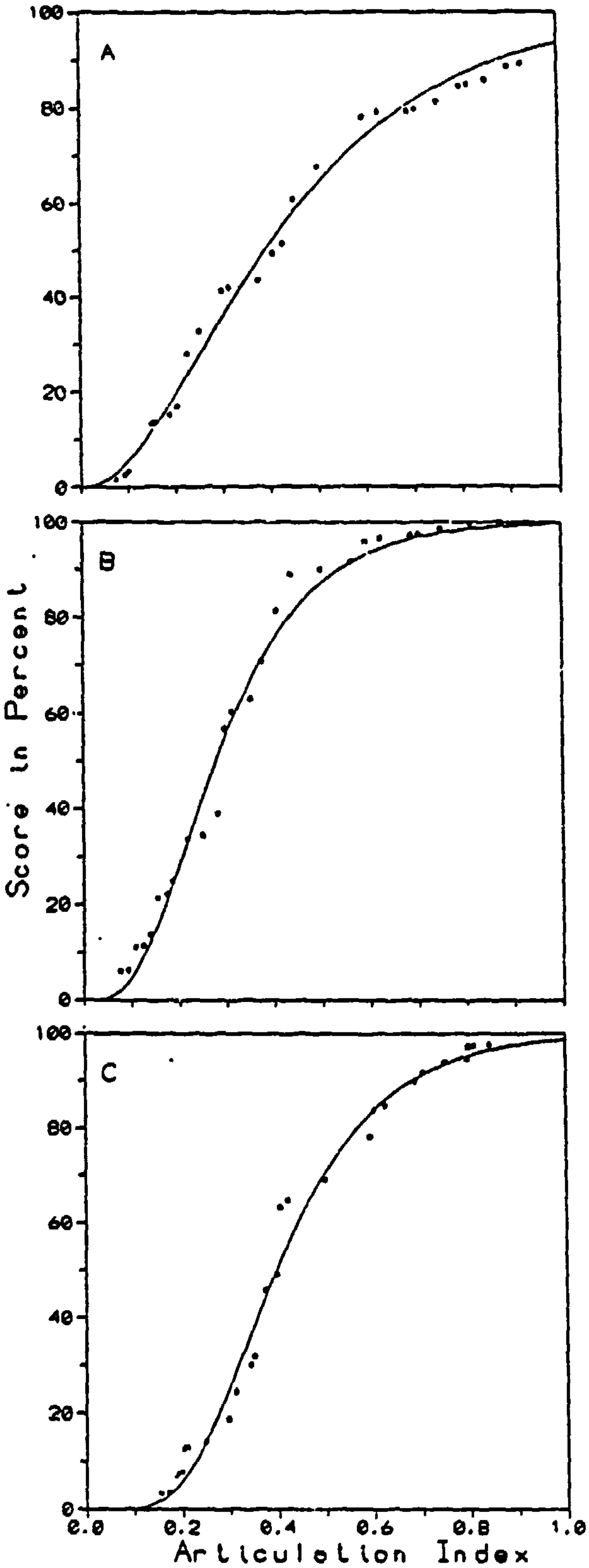


Figure 6: WRSs, SISs and CDISs vs. S/N ratio for sentences, words and CD.



<u>Figure 7</u>: Transfer function relating AI to articulation score, for a) words, b) sentences and c) CD. The crosses are data points.

as halving and complementing. The following is an example of each technique, using the subjects' WRSs as shown in Figure 6.

Halving

A basic assumption for determinig the AI is that the speech spectrum can be divided into frequency bands that are additive. Further, it is assummed that the crossover frequency divides the available speech spectrum or acoustic information into two equal halves. However, it should be noted that the S/N ratio and filtering condition which resulted in the highest WRS, SIS and CDIS did not truly deliver all of the acoustic information to each subject. This occurred because some noise was always present even at the most favorable S/N ratio and the response of the TDH-39 earphones limited the frequency output. Thus, instead of the generic term AI that implies that all of the acoustic information is available to the listener, the term AI_{max} will be used. Operationally defined, AI_{max} is the maximum obtainable articulation index using each speech type with the most favorable S/N ratio in the all-pass condition.

The following example illustrates the halving procedure which was used to derive the relative transfer function for the WRSs. Figure 8 shows the subjects mean transformed WRSs at a S/N ratio of +9 dB for each cutoff frequency filtering condition. The crossover frequency shown in Figure 8 corresponds to a WRS of 67.5% which is equal to $\frac{1}{2}$ AI_{max}. This WRS was the first one used to derive the WRS relative transfer function as is shown in Figure 7. The WRS of 67.5% also corresponded to a certain S/N ratio for words as shown in Figure 6 which is the amount of noise that must be added to the words to degrade the WRS to 67.5%.

Two steps were then performed to derive $\frac{1}{2}$ AI_{max}. First, the S/N ratio that degraded the words to $\frac{1}{2}$ AI_{max} (WRS=67.5%) was determined from Figure 6 which has been

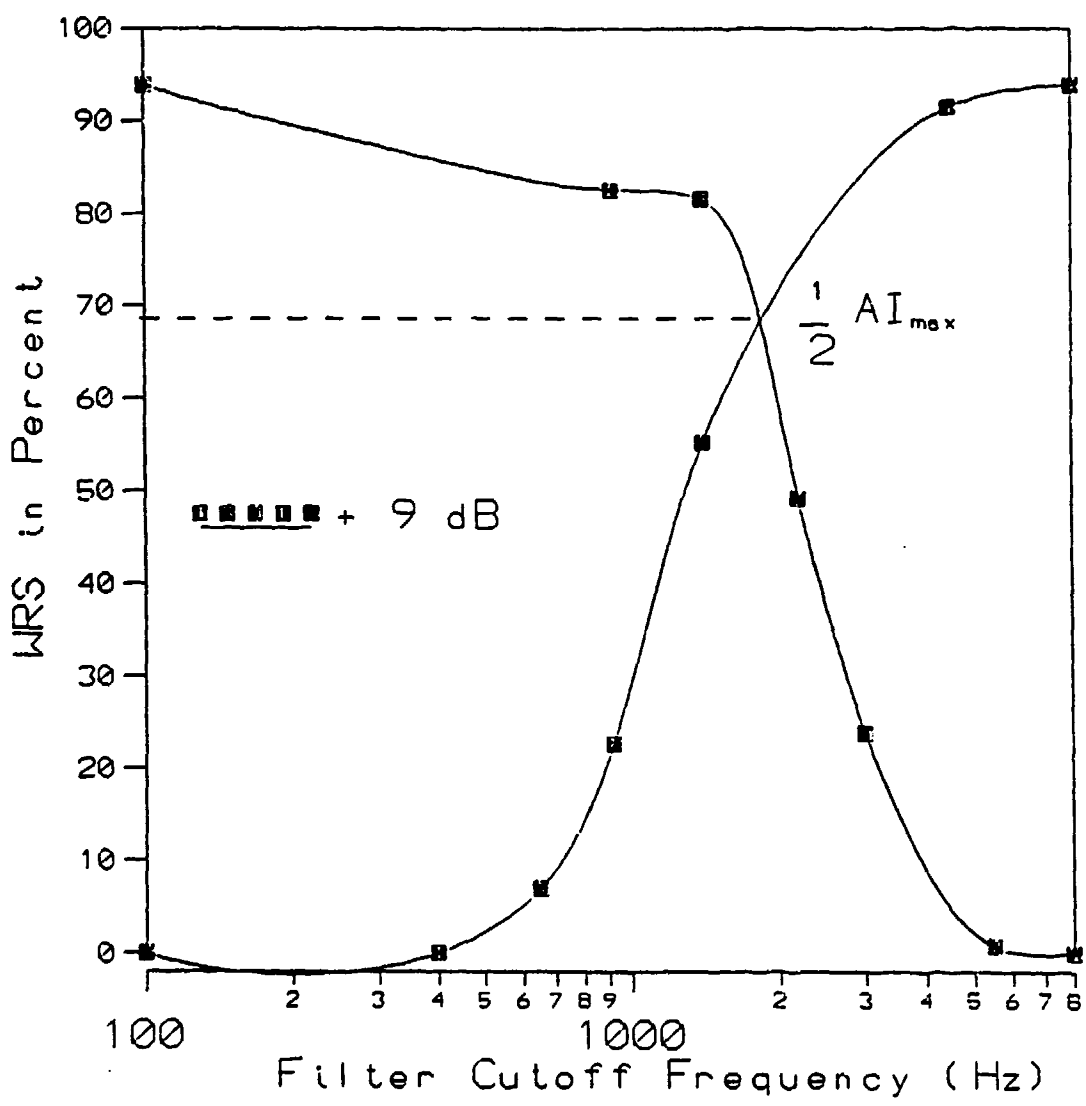


Figure 8: The HP and LP curves intersect at a WRS which corresponds to ½ AI_{max}. These curves are for words at +9 dB S/N.

reproduced in Figure 9a Inspection of Figure 9a reveals that a WRS of 67.5% corresponds to 0 dB S/N ratio. Second, the WRSs versus filter cutoff were plotted at a S/N ratio of 0 dB as shown in Figure 9b. The WRS corresponding to the crossover point was denoted as $\frac{1}{2}$ AI_{max} and was 32%. Thus, the WRS of 32% can also be considered $\frac{1}{2}$ of $\frac{1}{2}$ AI_{max}.

The next point derived by the halving procedure for the relative transfer function was $\frac{1}{2}$ of $\frac{1}{2}$ AI_{max}. This point was derived exactly like $\frac{1}{2}$ AI_{max}. The halving continued until the S/N ratio needed to degrade the speech signal to a WRS is smaller than the lowest S/N ratio used in this study was determined.

It should be noted that the S/N ratio obtained from Figure 9a was an arbitrary value. The smoothed curves for the subjects WRSs, SISs and CDISs for each S/N ratio and filter cutoff frequency did not provide enough information to derive any more points on the relative transfer function other than $\frac{1}{2}$ AI_{max}. This occurred because the WRSs, SISs and CDISs were obtained at five discrete HP and LP filter cutoff frequencies and one all-pass filter for each S/N ratio. As such, it was assumed that the distribution of WRSs, SISs and CDISs were linear between each actual score. Thus, additional data points (scores) were interpolated linearly between the actual scores. This was done in 1 dB increments for the WRSs and $\frac{1}{2}$ dB increments for the SISs and CDISs.

Complementing

Additional points on the relative transfer function were derived by complementing. Complementing takes advantage of the assumption that acoustic information in adjoining frequency bands is additive. Figure 10 illustrates the complementing procedure for the words using the +9 dB S/N ratio curve. A WRS of 32%, previously derived for $\frac{1}{2}$ AI_{max}, corresponded to two frequencies; one for the LP and one for the HP WRS curve. Since the AI is additive, the LP and HP frequencies that had a WRS of 32% intersect their

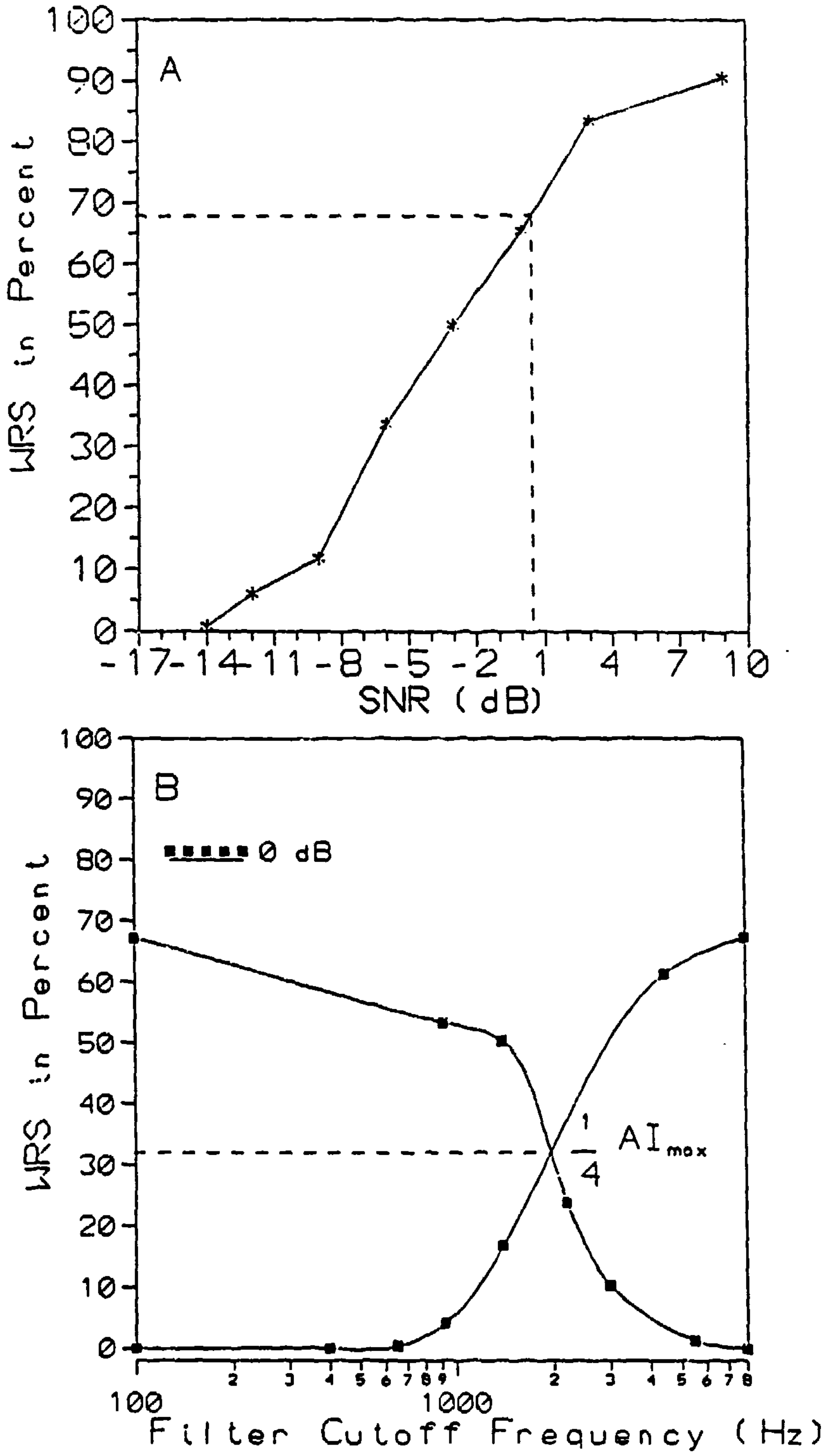


Figure 9: Two steps to perform halving. a) The WRSs vs. S/N ratio function is used to determine the S/N and b) the filter cutoff vs. WRS is plotted for that S/N. The intersection of these curves is $\frac{1}{4}$ AI_{max}.

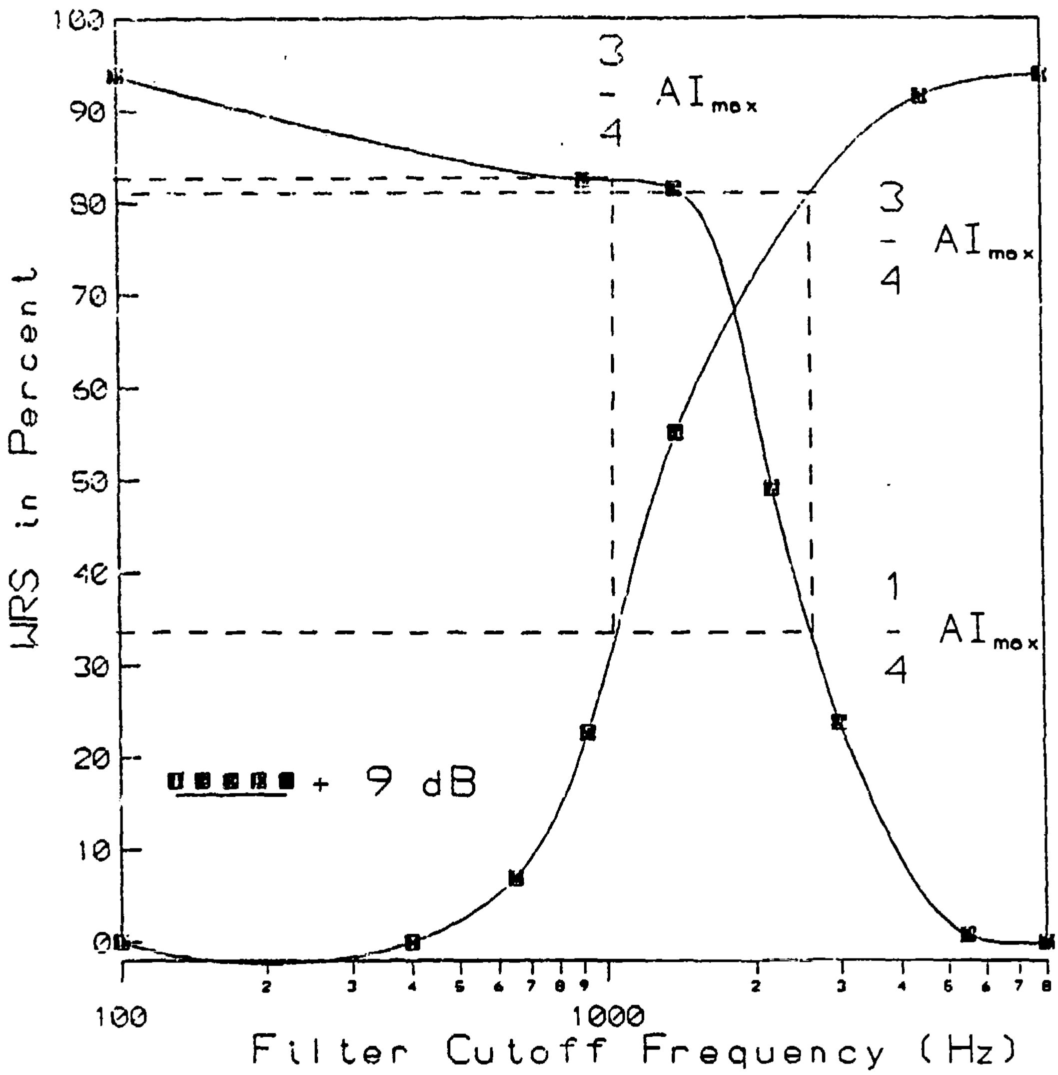


Figure 10: The WRS of 32% which corresponds to $\frac{1}{4}$ AI_{max} intersects the HP and LP curves and is then extended up toprodue two estimates for $\frac{3}{4}$ AI_{max}.

complementary HP and LP curve at WRSs corresponding to $\frac{3}{4}$ AI_{max}. Stated differently, a WRS of 32% corresponded to one LP and one HP filter cutoff frequency. Each of these frequencies intersects with a complementary HP or LP curve cutoff frequency so that a WRS corresponding to $\frac{3}{4}$ AI_{max} can be obtained. In each case, two WRSs were obtained at $\frac{3}{4}$ AI_{max} and were averaged to yield an AI value.

The complementing procedure can be performed on any of the points which are derived from the halving procedure and vice versa. For example, the halving procedure can be performed on the $\frac{3}{4}$ AI_{max} to obtain .375 AI_{max}. The .375 AI_{max} is then complemented to yield .1875 AI_{max}.

Fitting the curve

Overall, the halving and complementing procedures were repeated until 26 points were derived so that a relative transfer function for the words, sentences and CD passages could be defined. The curve that best fit the relative transfer function data was introduced by Fletcher and Galt (1950), recently advocated by Studebaker and Sherbecoe (1991), and is shown below as:

$$S = \left(1-10^{\frac{(-AI)(P)}{Q}}\right)^{N} \qquad (11)$$

In equation 11, S is the WRS, SIS or CDIS and P is the proficiency factor which is the proficiency of the talker and listener combination and was assumed to be one. The fitting constants for Q and N as well as R^2 (coeficient of determination which is a measure of the goodness of fit of the data to equation 11) are shown in Table 9. The curve was fit to the data using the SAS NLIN procedure (SAS, 1985). The curves or relative transfer functions derived from equation 11 for each type of speech are shown in Figure 6.

Table 9: Fitting constants for three types of speech using equation 11

Speech Type	Q	N	R^2
words	0.6408773	2.4355182	0.986
sentences	0.3289019	4.4807825	0.987
CD	0.3525916	8.9429450	0.985

Derivation of the Frequency Importance Function

A frequency importance function defines the relative importance of frequency bands in the speech spectrum contributing to the intelligibility of speech. Figure 11 shows the frequency importance functions for the words, sentences and CDs used in this study. Each frequency importance function was obtained using the transformed mean WRSs, SISs and CDISs obtained in the five LP and HP pass filter conditions at each S/N ratio previously shown in Figures 3, 4 and 5 respectively. In addition, the WRSs, SISs and CDISs obtained in the all-pass condition for each S/N ratio and zero performance scores were also used. Finally, three more WRSs, SISs and CDISs were used to derive the frequency importance functions using the following procedure.

Of the five LP and five HP filter cutoff frequencies, the only filter cutoff frequencies common to both LP and HP were 920 and 1400 Hz. The three other LP (400, 600 and 4500) and HP (2200, 3000 and 5500) filter cutoff frequencies were paired using the following procedure. An example of the procedure for the words is illustrated in Figure 12. Specifically, Figure 12 shows the transform mean WRSs obtained for the LP and HP filter cutoff frequencies at a S/N ratio of +3 dB. This data was shown previously in Figure 3 and Table 6. The asterisks shown in Figure 12 were derived from the smoothed curve of the subjects WRSs as shown in Figure 3. The frequency corresponding to each asterisk

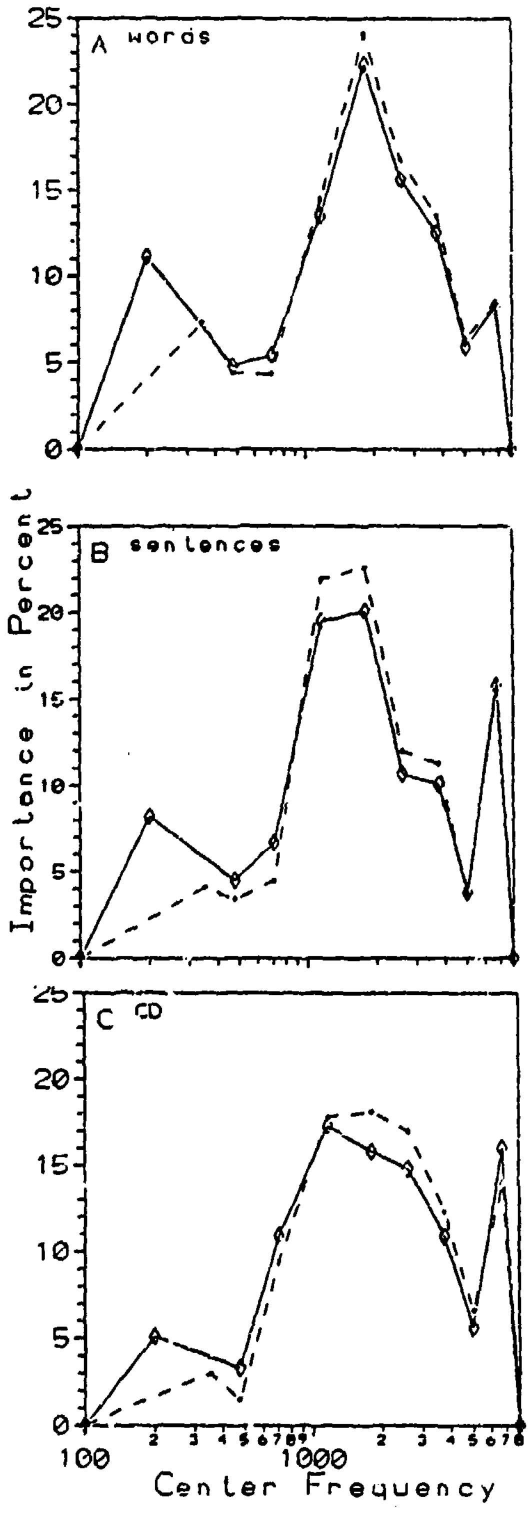
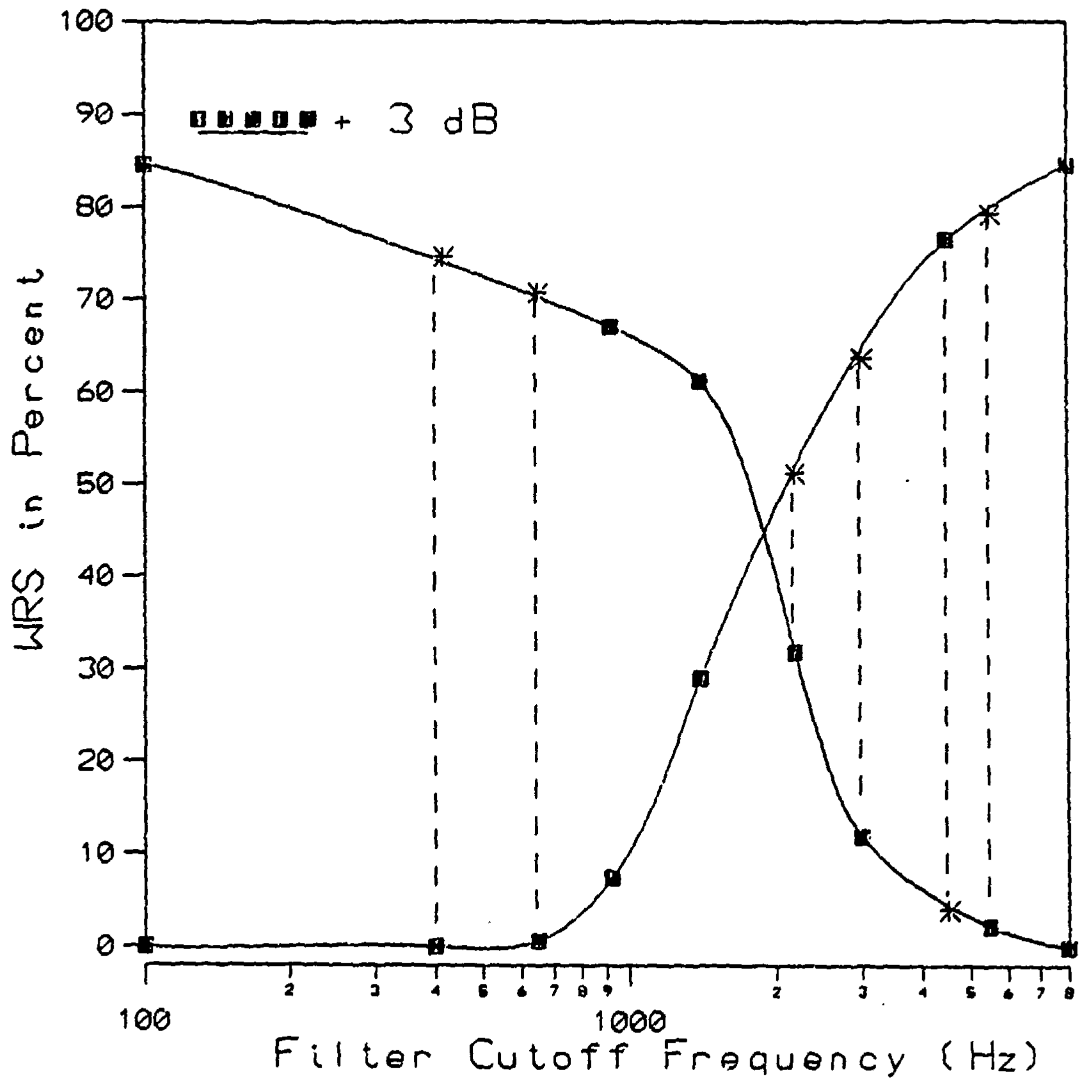


Figure 11: Frequency importance functions for a) words b) sentences and c) CD with zeros included (dashed) and excluded (solid).



<u>Figure 12</u>: Example of a pair of modified curves which were used to derive the frequency importance function (words, ÷3 dB S/N ratio). The dashed lines begin at a data point and end at a derived band estimate (asterisk).

corresponds to one of the LP or HP filter cutoff frequencies that did not have a matching filter pair (the dashed lines illustrate this).

To obtain band estimates, each WRS shown in Figure 3 for each filter cutoff frequency was converted from percent to an AI score using equation 11 and the fitting constants shown in Table 9. The AI differences between two contiguous cutoff frequencies in either the HP or LP filter curve was taken as an estimate of the frequency importance of that band. More specifically, band estimates for the LP filter cutoff frequencies were obtained by subtracting the AI value for the lower LP cutoff frequency from the band with the higher LP cutoff frequency. The reverse was used for high pass filters. These band estimates are presented in Tables 10, 11 and 12.

Inspection of Tables 10, 11 and 12 reveals that in some cases the band estimate was negative and were assumed to be zero. Studebaker and Sherbecoe (1991) did not consider negative or zero band estimates as data. Instead, they used a zero bands pair estimate (each SNR yielded a pair of band estimates, one HP and one LP) as the lone estimate of the frequeny importance of that band at that particular S/N ratio. Although no explanation was given, possibly Studebaker and Sherbecoe (1991) were of the opinion that band estimates of zero did not demonstrate that listeners were unable to use the available acoustic information, but rather they may have thought that scme condition within the experimental paradigm (such as excessive masking) prevented the listener from using the available acoustic information. This issue will be explored in the next chapter. However, both methods of treating negative or zero band estimates are included here.

Figure 11 shows the frequency importance functions for each speech type. The dotted lines represent the frequency importance functions with the inclusion of zero and the solid lines for the exclusion of zero band estimates. When computing the band estimates, if the zero band estimates were not included, the estimate for the frequency importance of the

Table 10: Band estimates for words

	BAND ESTIMATES							
	-9 dB		-3 dB		+3 dB		+9 dB	
Band	Low	High	Low	High	Low	High	Low	High
100-400	.000	-.010	.000	.048	.007	.155	.027	.224
400-650	.000	.008	.012	.014	.030	.044	.085	.044
650-920	.000	.000	.028	.004	.079	.028	.105	.015
920-1400	.007	.074	.049	.001	.139	.052	.206	.016
1400-2200	.029	.019	.106	.143	.145	.200	.173	.316
2200-3000	.017	.039	.091	.049	.097	.123	.144	.157
3000-4500	.023	.002	.073	.061	.128	.057	.183	.126
4500-5500	.016	.007	.011	.035	.049	.026	.072	.059
5500-8000	.036	.012	.013	.029	.076	.065	.000	.039

Table 11: Band estimates for sentences

	BAND ESTIMATES							
	-12 dB		-8 dB		-4 dB		+0 dB	
Band	Low	High	Low	High	Low	High	Low	High
100-400	.000	.000	.000	.040	.000	.104	.065	.000
400-650	.000	.000	.000	.012	.063	.058	.044	.000
650-920	.000	.000	.036	.008	.039	.031	.101	.000
920-1400	.080	.037	.090	.011	.176	.078	.257	.184
1400-2200	.028	.047	.061	.102	.120	.274	.220	.430
2200-3000	.016	.034	.036	.059	.087	.083	.142	.151
3000-4500	.019	.018	.047	.017	.150	.045	.170	.071
4500-5500	.010	.000	.022	.007	.076	.022	.000	.036
5500-8000	.025	.050	.042	.077	.088	.105	.000	.127

Table 12: Band estimates for CD

	BAND ESTIMATES							
	-12 dB		-9 dB		-6 dB		-3 dB	
Band	Low	High	Low	High	Low	High	Low	High
100-400	.000	.012	.000	.018	.000	-.040	.070	.079
400-650	.000	.000	.000	.009	.000	-.010	.082	.013
650-920	.000	.007	.115	.013	.190	-.010	.099	.025
920-1400	.110	.020	-.020	.059	.023	.335	.136	.154
1400-2200	.054	.032	.104	.060	.092	.108	.144	.266
2200-3000	.024	.079	.043	.070	.121	.155	.125	.165
3000-4500	.028	.012	.047	.037	.152	.041	.291	.054
4500-5500	.016	.000	.025	.098	.047	.018	.029	.026
5500-8000	.034	.097	.055	.000	.087	135	-.010	.179

band at a particular S/N ratio was based solely upon a HP or LP filter. If both estimates were zero, no band estimate at that S/N ratio was used. The band estimates were normalized to one within each S/N ratio and then averaged across all four S/N ratios. Finally, the band's total importance was determined in percent.

Derivation of the Absolute Transfer Functions

The relative transfer functions were adjusted to reflect the total amount of information available to the subjects. This was done as follows. Equation 12 was used to estimate the total amount of information available at each S/N ratio. Equation 12 is a modification of equation 5 and reflects the fact that the numerator is simply the speech to noise (SP/N) ratio.

$$W_n = \left(\frac{SP/N}{30}\right) \qquad (12)$$

It is important not to confuse the SP/N ratio with the S/N ratio. The SP/N ratio is the level in dB above that point at which the subjects WRS, SIS or CDIS is zero (0 dB SP/N ratio).

Recall from equation 5 that W_n is the weighting factor which quantifies how much of the acoustic information is available to the listener due to noise. Further, the product of the weighting factor with the frequency importance function yields the AI. In the all-pass condition, the AI can be estimated from equation 12, since the frequency importance function will add to one. However, the estimate will only be as accurate as the estimate for 0 dB SP/N ratio.

Another method for obtaining an estimate for AI is to solve equation 11 for AI; this yields:

$$AI = -Q\log(1-s^{\frac{1}{N}}) \tag{13}$$

Equation 13's estimate of AI is based upon the derived points that relate the AI to the subjects WRSs, SISs and CDISs. However, the use of equation 13 requires an estimate of AI_{max} so that the scores used to derive the relative transfer function may be scaled to reflect the fact that AI_{max} was not equal to one. Equation 12 provides a true AI_{max} if the estimate for 0 dB SP/N is accurate. Thus, the difference between the point estimated by the theoretical curve (which best fit the data) and the estimate based upon the level of the speech above the noise, should be similar provided that the 0 dB SP/N is accurate. On the other hand, if this estimate for 0 dB SP/N ratio is incorrect, the difference between equations 12 and 13 at any of the all-pass conditions in Figure 6 should be large.

Thus, the absolute transfer functions were derived by minimizing the sum of the absolute value of the difference between equations 13 and 12. as shown in equation 14.

$$AI_{(min)} = \sum_{SNR=1}^{8} \left| -Q\log(1-s^{\frac{1}{N}}) - \left\{ -\frac{SP/N}{30} \right\} \right| \tag{14}$$

The left side of the difference in equation 14 is the fitted curve. The right side is the predicted AI value based upon the amount of speech information which is present above the noise (Beranek, 1947 and Kryter, 1962a). The dynamic range of speech was assumed to be 30 dB (Beranek, 1947 and Pavlovic and Studebaker, 1984).

In order to calculate the AI values using equation 14, an estimate for 0 dB SP/N (0 dB SP/N is the S/N ratio at which the articulation score is 0%) was required. This was needed, not only to get an estimate for the right side of the difference in equation 14, but also in order to find values for Q and N (fitting constants) for the right side. The value that minimized equation 14 was found by iteratively varying the assumed 0 dB SP/N estimate. For each value new fitting constants were calculated and a new value for equation 14 was computed. The difference between the estimates of 0 dB SP/N was halved until four decimal place accuracy was achieved. The 0 dB SP/N ratios were -12.0625 for words, -12.7500 for sentences, and -15.6250 for CD. The absolute transfer functions are illustrated in Figure 13 assuming the appropriate AI_{max}.

Inspection of Figure 13 reveals that the slope of the absolute transfer function for words is very different from that of both sentences and CD. In effect, as more acoustic information is available to the listener, the WRSs increase at a slower rate than both the SISs and CDISs.

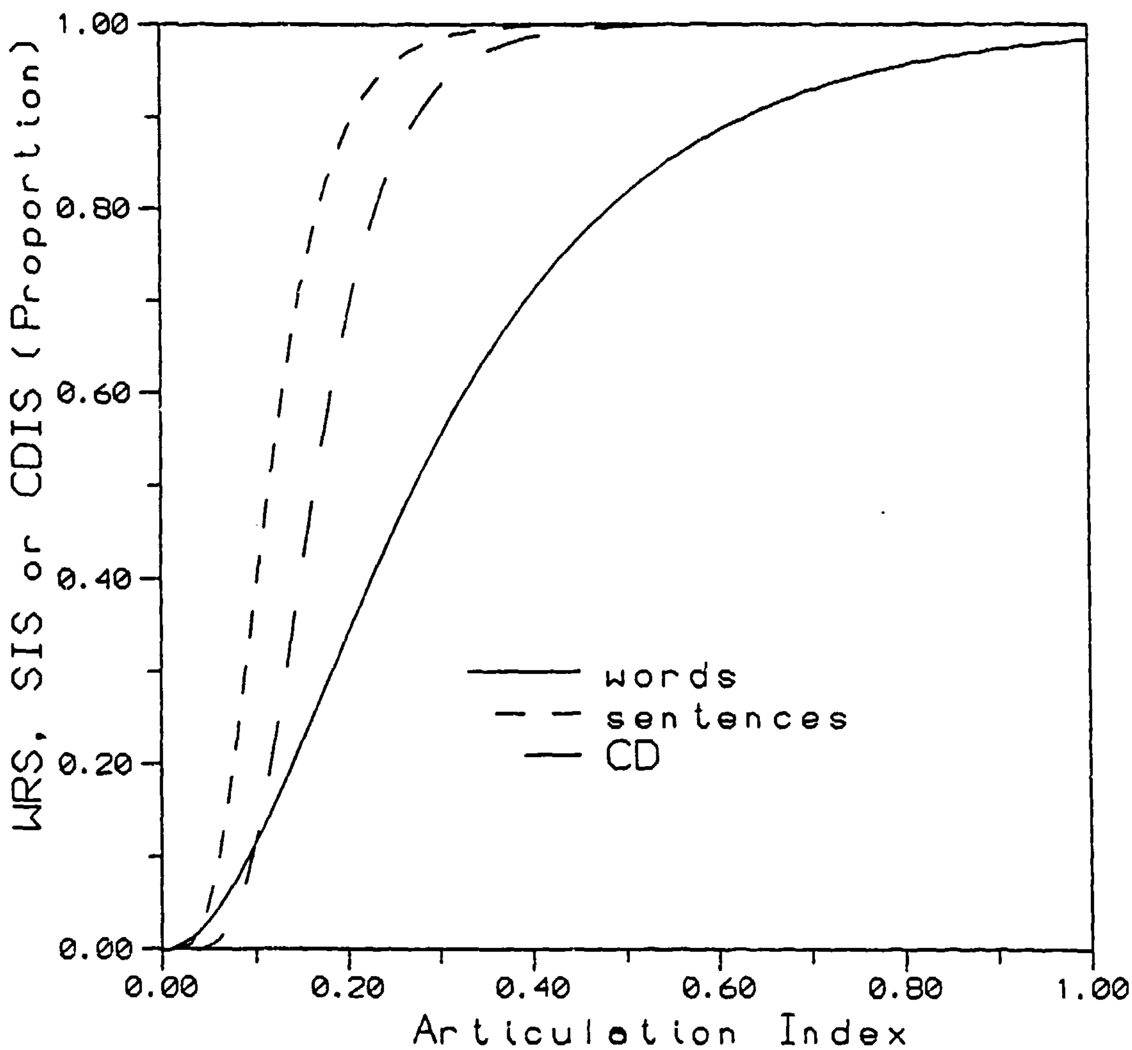


Figure 13: Absolute transfer functions for words, sentences and CD

Chapter 6

DISCUSSION

Overall, the purpose of this research was to determine the intelligibility of words, sentences and CD for several LP and HP filter cutoff frequencies in several S/N ratios. Unlike previous research, in the present study all testing for each type of speech was conducted with the same subjects (N=24), one talker, identical instrumentation, and extremely well controlled conditions. Typically, previous research research has been confounded by differences in methodology, assumptions, subjects, instrumentation and talkers. As such, a unique aspect of the present study was that all of the results across speech type were free from any differences that existed in previous research. The experimental objectives were to

1. Derive transfer functions relating the AI to WRSs, SISs, and CDISs;
2. Compare transfer functions for a word identification task to a sentence and CD estimation task; and
3. Develop frequency importance functions for words, sentences and CD.

The Transfer Functions for Words, Sentences and CD

The first experimental objective was to develop transfer functions relating the AI to words, sentences and CD. The relative transfer functions were derived for each speech type from the curves in Figures 3, 4 and 5. Each transfer function was based upon 26 points. The WRS data points ranged from 2.5% to 89% or an AI ranging from .074 to .926, while the SISs ranged from 6% to 99.83% or an AI from .078 to .875 and for the CDISs from 3.25% to 97.58% or an AI from .156 to .844. The wide range of WRSs, SISs and CDISs derived from the four S/N ratios signified that the spacing between the highest

and lowest S/N ratio was sufficient to adequately define the transfer function. The lowest and highest S/N ratios limited the lower and upper AI values that could be derived from the transfer functions. This occured because the highest %/AI point was derived by the complementing procedure using the highest S/N ratio while the lowest %/AI was derived by the halving procedure using the lowest S/N ratio.

Comparison of the Transfer Functions

The second experimental objective was to compare the transfer functions derived for words, sentences and CD. One unique aspect of this study was that comparisons between speech types could be made with confidence that any disparities between words, sentences and CD passages were actually due to differences in the stimuli. That is, between the speech types there were no intervening variables except the inherent difference in message redundancy.

The first point of comparison concerns the 0 dB SP/N ratio values. These values were important because they determined AI_{max} for each type of speech. Further, These values were used to scale each AI value in the relative transfer functions in order to determine the absolute transfer function. The scaling constant was AI_{max} divided by 30 dB which was the assumed dynamic range of speech. The values for 0 dB SP/N ratio were -12.0625 for words, -12.75 for sentences and -15.625 for CD. It was interesting that there was such a small difference between the 0 dB SP/N ratio for words and sentences. Contextually, the sentences were more similar to the CD passages than the words. The comparable 0 dB SP/N ratio for sentences and words might be due to the stimulus length. The sentences were four to five seconds in length versus the CD Passages, which were 20 to 25 seconds long. It seems likely that the subjects would be more successful identifying the subject

matter of the CD passages in unfavorable filter and noise conditions than for the sentences simply due to the increase in the stimulus length. If so, the subjects' estimates would be considerably higher for unfavorable noise conditions, since the estimates of percentage of words understood for CD (CDIS) would be judged higher due to the knowledge of the subject matter of the passage versus the sentence estimates (SISs) based upon only the number of words heard, with no knowledge of the meaning of the se

Alternatively the difference in 0 dB SP/N ratio scores could be related to the extreme difficulty in hearing even one word in an average four to five second sentence. Therefore, the chances of identifying the subject matter of the sentence in highly degraded conditions was disproportionately lower for the sentences than for the CD passages which were 20 to 25 seconds. Thus, the subjects use of the semantic and syntactic cues present in the sentences would be considerably lower for the most severe filtering and S/N ratios than for CD. This also would explain the comparable steep slopes of the curves in Figure 6 for the sentences and CD in spite of the large difference between their 0 dB SP/N ratios. As the degrading of the speech become less severe, the SISs amd CDISs become similar.

The second point of comparison was the transfer function for a word identification task to the sentence and CD estimation task. Table 13 shows difference scores (SIS-WRS, SIS-CDIS and CDIS-WRS) between the curves in Figure 13 for .05 AI increments. Inspection of Table 13 reveals that the differences between the SISs-WRSs and CDISs-WRSs is large and occurs between the .1 and .65 values of AI. Since the difference scores were as large as 54.4% for the SIS-WRS and 39.4% for the CDIS-SIS, the absolute transfer function for words was not comparable to the absolute transfer function for sentences and CD. The difference scores for the SISs-CDISs were more interesting. The largest differences occur from the .1 to .25 AI values and were 26.8, 30.5, 20.3 and 11% respectively. If the absolute transfer function for the SISs or CDISs were displaced a .05

AI value toward the other, the SISs-CDISs differences would change to 6.8, 2.5, -3.2 and -3.9% respectively.

There are many factors which might shift the transfer function for sentences and CD. For example, a 1.5 dB error in the estimation of the 0 dB SP/N ratio for either the SISs or CDISs would shift the respective transfer function 0.05 AI. In this study, it is questionable whether an AI difference of 0.05 is significant.

The similarity of the sentence and CD transfer functions might be significant. The control of factors that affect intelligibility are very difficult for CD (Giolas, 1966, Fry, 1968). Syntactic and semantic factors are much easier to control in sentences (Kalikow et al., 1977, Giolas, 1970) and so, the use of sentences in place of CD might be beneficial in some situations. For example, one method to determine the acoustic environment needed to achieve a certain level of intelligibility for connected speech would be to perform extensive word recognition testing. Although this method allows for the exact specification of the speech stimulus (word length, phonetic balancing, word frequency) it would only be an indirect measure of the intelligibility of connected speech. If 30% of the acoustic information would be available to the listener (AI=0.30), this would correspond to a WRS of 55.1% and a SIS of 98.4% (Table 13). The WRS severely underestimates the intelligibility of connected speech since the CDIS for AI equal to 0.30 was 92.9%. The SIS was very close to the CDIS and thus might be used in place of the words to obtain a more accurate prediction of the intelligibility of connected speech in a noisy environment.

The Frequency Importance Functions

The third experimental objective was to derive the frequency importance functions for words, sentences and CD. The frequency importance function (I(f)) is a plot of the relative

Table 13: Difference scores for WRSs, SISs and CDISs in percent. The value represent the difference between the percentage scores with the AI value held constant.

	DIFFERENCE SCORES		
AI	SIS - WRS	SIS - CDIS	CDIS - WRS
0.05	1.7	4.5	-2.8
0.10	26.9	26.8	0.1
0.15	48.9	30.5	18.4
0.20	54.4	20.3	34.1
0.25	50.4	11.0	39.4
0.30	43.3	5.5	37.8
0.35	35.9	2.6	33.3
0.40	29.0	1.1	27.9
0.45	23.3	0.5	22.8
0.50	18.6	0.3	18.3
0.55	14.8	0.0	14.8
0.60	11.7	0.0	11.7
0.65	9.2	0.0	9.2
0.70	7.3	0.0	7.3
0.75	5.7	0.0	5.7
0.80	4.5	0.0	4.5
0.85	3.6	0.0	3.6
0.90	3.4	0.0	3.4
0.95	2.2	0.0	2.2
1.00	1.7	0.0	1.7

importance of the bands of speech that were filtered in this experiment to the intelligibility of speech. The accuracy of I(f) depends upon the number of filters and S/N ratios.

This study used 11 filter conditions that provided frequency importance estimates at nine frequency bands. There were 4 S/N raios that provided eight (one HP and one LP estimate for each S/N ratio) frequency importance estimates for each of the nine frequency bands. One disadvantage to using only 4 S/N ratios was that extreme estimates had a larger influence upon the average. This was noticeable in the highest frequency band (5500-8000 Hz). One sentence frequency importance estimate in this band was .127, or .39 from the next highest estimate. The highest CD estimate (.179) was .44 from the next closest estimate. Since both of these high estimates occured in the HP condition in the least severe S/N ratio (-3 and 0 dB for CD and sentences respectively) they might simply demonstrate that the listeners used more high frequency information as the level of the noise decreased. Even so, they had a large influence on the importance of the frequency band of 5500-8000. The same effect was noticeably for the lowest frequency band (100-400 Hz).

The high frequency emphasis in the importance functions deserved closer inspection. There is a trend in the band estimates for both the sentences and CD passages. The actual high frequency band estimates (the low frequency band estimates for the 5500-8000 Hz band are derived estimates) increase as the S/N ratio became more favorable. One possible explanation is that the FIR filters had a slope of 60 dB per octave. At lower frequencies this was not an issue since, for example, a 60 dB per octave filter with a cutoff frequency at 1000 Hz is attenuated 60 dB at 500 Hz. Unfortunately, a 5500 Hz filter cutttoff frequency will not attenuate 60 dB until 2750 Hz. Thus, it would be expected that for more favorable S/N ratios some information below 5500 Hz could affect the subjects' estimates. This, coupled with the small number of S/N ratios used in this experiment, undoubtedly caused high estimates of the highest frequency band.

The use or non use of estimates that are either zero or negative is another important issue. Ignoring these estimates assumes that the subject is not able to use this information, possibly due to excessive masking. At high noise levels there also might be loss of acoustic information due to the spread of masking. In this experiment, the noise levels ranged from 64 to 85 dB SPL. At the highest levels (85 and 82 dB SPL), there could have been some upward spread of masking; however, it would be minimal (Kryter, 1985 and ANSI S3.5-1969). Also, since the noise was shaped to within ± 1 dB of the one-third octave long term rms speakers spectrum, it was assumed that there was no excessive masking at any one frequency band. For this reason, all estimates of either zero or negative values were considered valid data. The use of these values would imply that the subjects were supplied with this acoustic information but were not able to use it.

Figure 14 illustrates the I(f) for words, sentences and CD including zero and negative band estimates. As the message redundancy increased (words to sentences to CD) the area of most importance changed from a sharp peak (24%) at the band centered around 1800 Hz for words to a less dramatic peak (~ 18%) encompassing the bands centered around 1160, 1800 and 2600 Hz for CD. The sentences were in between, with a sharp peak (~ 22%) at the bands centered around 1160 and 1800 Hz. Two points can be made concerning the strategy of the subjects in this study

1. As message redundancy increased, the consonantal cues (acoustic cues centered around 2000 Hz) become less important;

2. As message redundancy increased the shape of the I(f)s primary area of importance spread out to include more low frequency and more high frequency cues;

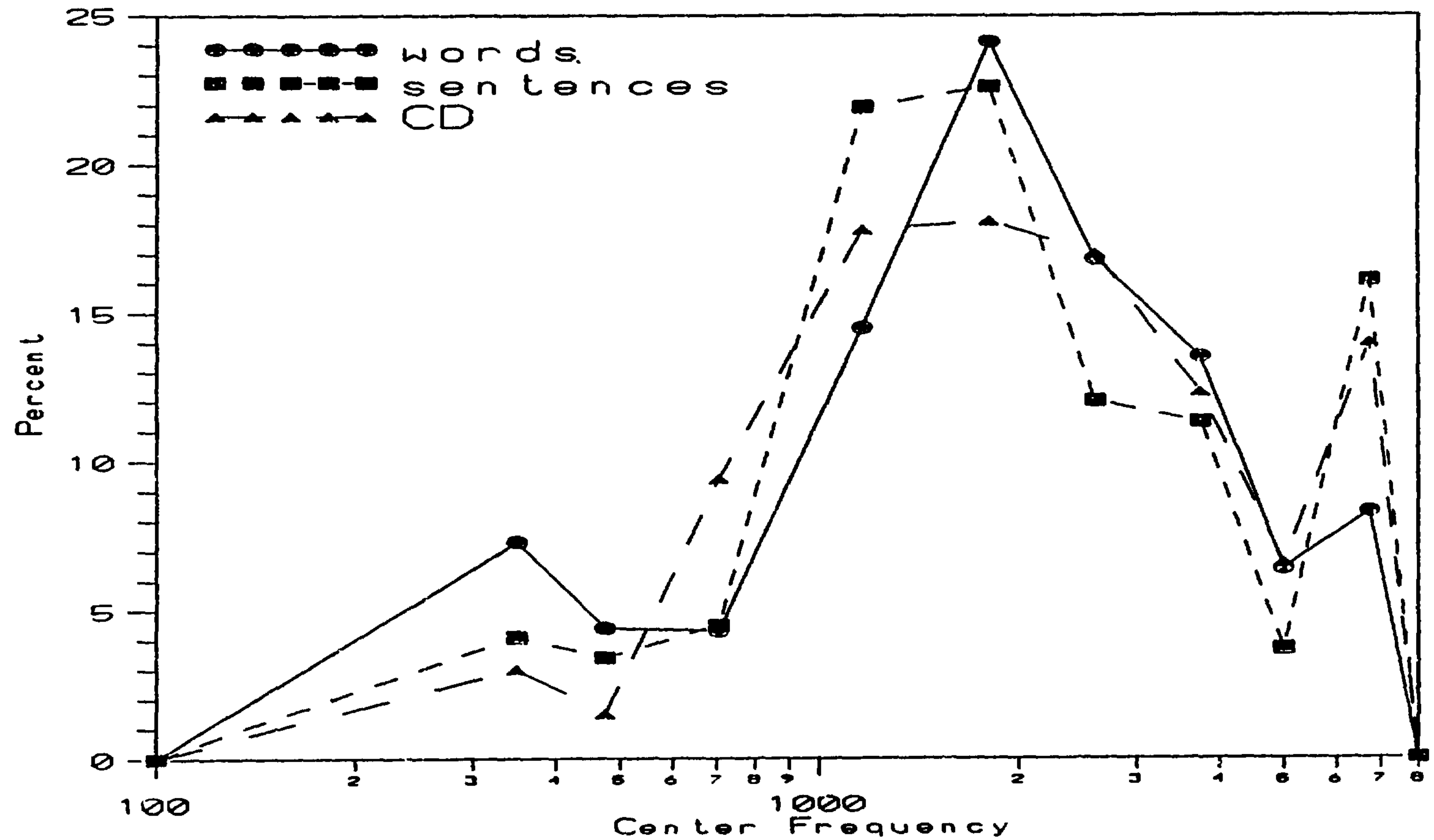


Figure 14: The frequency importance function for words, sentences and CD including zero and negative band estimates.

Comparative Analysis

A unique aspect of this research was that it incorporated a broad range of speech types having different amounts of message redundancy in a single study. Previous studies have concentrated upon just one type of speech (Studebaker et al., 1987, Black, 1959, Studebaker and Sherbecoe, 1991 and Schum et al., 1991). This following will compare the results of the present study with other research.

The Transfer Function

The AI has been used extensively to study the intelligibility of isolated words. Transfer functions have been developed for monosyllabic words (Black, 1959), NU-6 word lists (Schum et al., 1991) and the CID W-22 word lists (Studebaker and Sherbecoe, 1991). Figure 15 illustrates these transfer functions and findings in the present study for words.

Two points can be made concerning the transfer function provided by Black (1959), Schum et al. (1991) and from the present study. First, these studies all used monosyllabic words from the original set of PB-50 word lists. Since each study used words that were similar in word frequency, phonetic content and syllable length the transfer functions should have the same slope. Secondly, each study presented each word once to each subject. The words were all unknown to each listener. Thus, each curve should have similar word recognition scores for similar AI values.

The Studebaker and Sherbecoe (1991) transfer function for NU-6 words is the only transfer function that is noticeably different. This is also the only research that used a small number of subjects (eight) repeatedly listening to randomized versions of the same

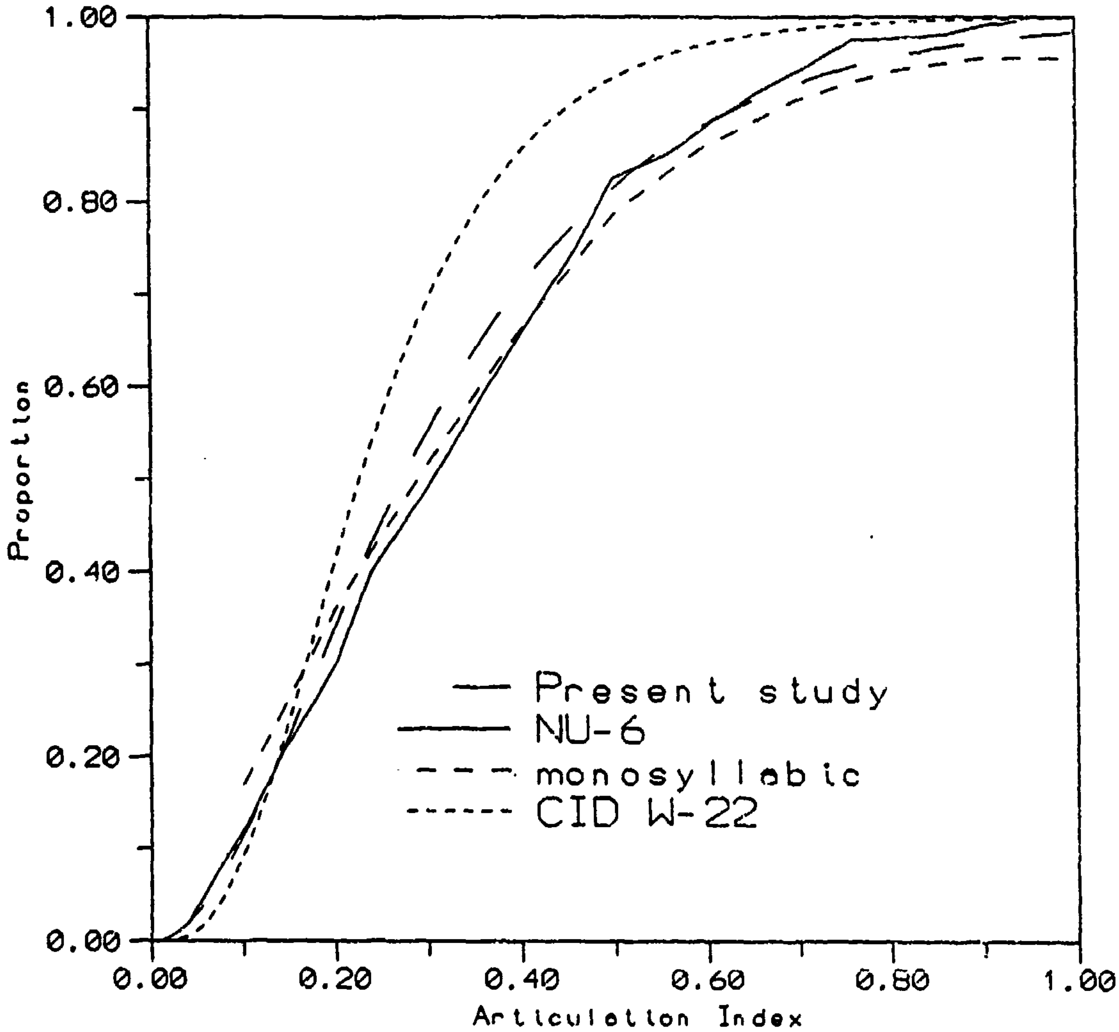


Figure 15: Transfer function derived for words in this study with comparable curves for monosyllabic words (Black, 1959), NU-6 word lists (Schum et al., 1991) and the CID W-22 word test (Studebaker and Sherbecoe, 1991).

four fifty word lists. Each subject was presented one randomized version of four 50 W-22 word lists at each of 308 filter and noise conditions. Thus, the subjects in the Studebaker and Sherbecoe (1991) study were highly trained and effectively received a limited vocabulary. French and Steinberg (1947) made some calculations, predicting the effect of limiting vocabulary size in intelligibility which were used in ANSI S3.5-1969. These relations are the ones previously shown in Figure 2. Inspection of Figure 2 reveals that the expected transfer function changed, as vocabulary was limited, steepening the slope (a greater change in the AI for the same change in %) and a shift of the position of the function to the left. This would signify that identical AI values would correspond to a higher percentage of words identified. This is exactly the relation between the Studebaker and Sherbecoe (1991) study (limited vocabulary and trained subjects) and the results of Black (1959), Schum et al. (1991), and the present study (large vocabulary and untrained subjects).

Previously, transfer functions have not been directly derived for meaningful sentences. Schum et al. (1991) derived a curve for the SPIN non-meaningful sentences (SPIN), and the results were identical to the curve in Figure 15 for their transfer function for the NU-6 word lists. This implies that a word recognition and non-meaningful sentence key-word in sentence identification task are identical with repect to the AI. The difference between the transfer functions for a sentences estimation task in this study and the key word identification task by Schum et al. (1991) (Figure 13 versus Figure 15) illustrates considerable difference due to the differences in the respective tasks and the difference in message redundancy.

Figure 16 illustrates the CD transfer function obtained in this study and the one reported by Studebaker et al. (1987). Both studies used passages approximating a 7th grade reading level which may account for the similarity in slope. Although it may seem

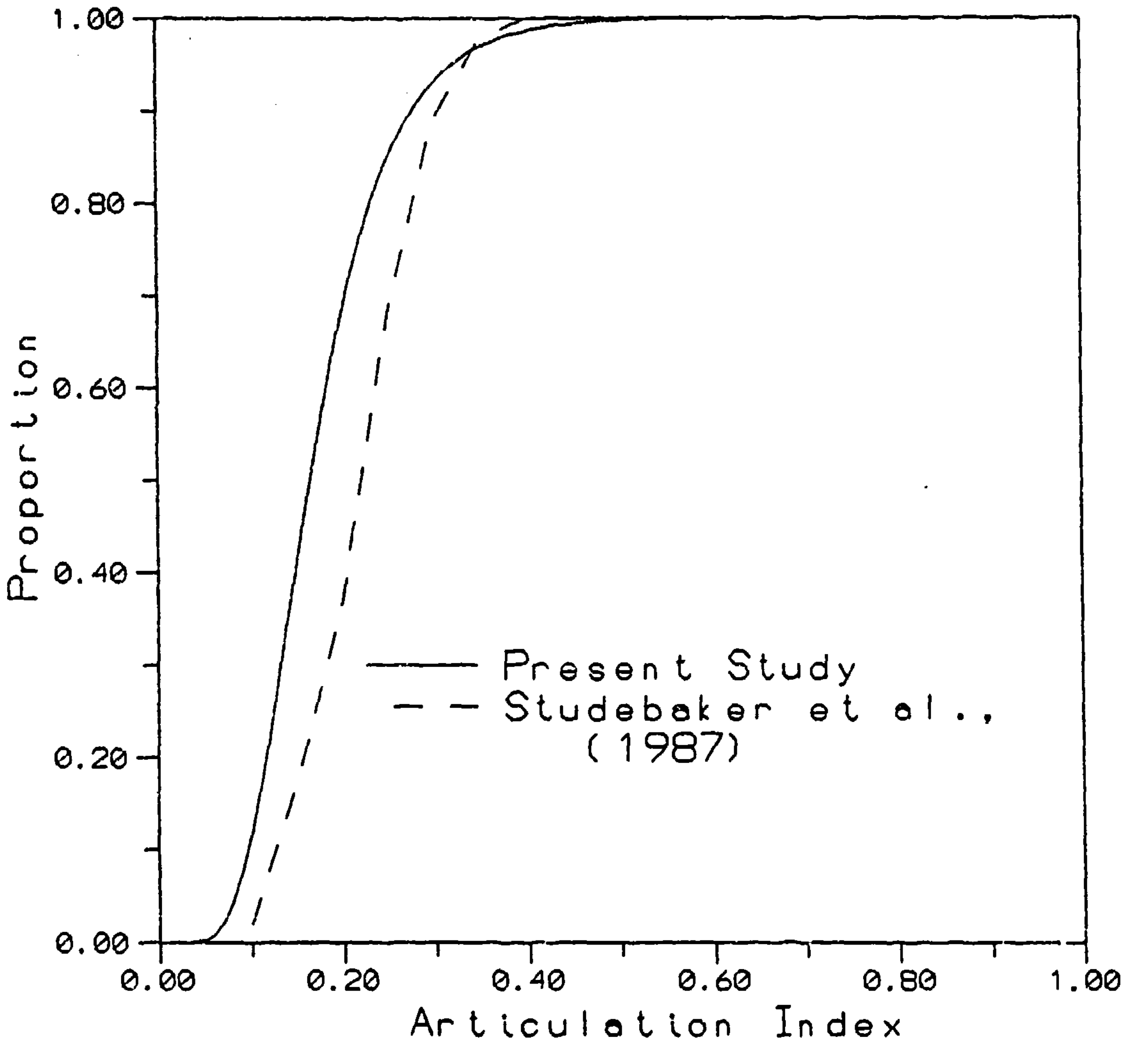


Figure 16: The transfer function derived for CD with a similar curve for CD by Studebaker et al., 1987.

that there is a difference between the two curves, care must be taken when considering small differences across studies. There is always a degree of difference across studies due to analysis techniques, signal measurement procedures and assumptions. Due to the limitations inherent between comparing data across studies, the small differences between the studies are probably not significant. All that can be stated with certainty is that the transfer function of Studebaker et al. (1987) versus and the one obtained in this study maintain the expected relation with the transfer function for words.

Frequency Importance Function

Figure 17 is a comparative plot of three I(f)s for words and the ANSI S3.5-1969 I(f) for nonsense syllables. All the data from the studies, other than the present one, were either developed in one-third octave bands (Schum et al., 1991 and Studebaker and Sherbecoe, 1991) or recalculated into one-third octave bands (Studebaker et al., 1987, and Black, 1959) by Studebaker (1987). The frequency importance estimates in the present study were not suitable for presentation in one-third octave bands. This was due to the nature of recalculating data from large frequency bands into smaller bands. Although the importance in percent is known (for example when computing the frequency importance of three one-third octave bands from one one octave band) the shape of the function is not unique. The reverse procedure is not problematic. Three one-third octave bands simply add to produce one estimate at the center frequency of the octave band in question. Therefore, the data from the past studies was recalculated into octave bands. The center frequencies were 125, 250, 500, 1000, 2000, 4000 and 8000 Hz. The frequency importance estimates for the present study were not recalculated into octave bands. since the band estimates in this study were close enough to octave bands to allow comparison. These frequency bands are listed in Table 14 beside the comparable octave bands used with the other studies.

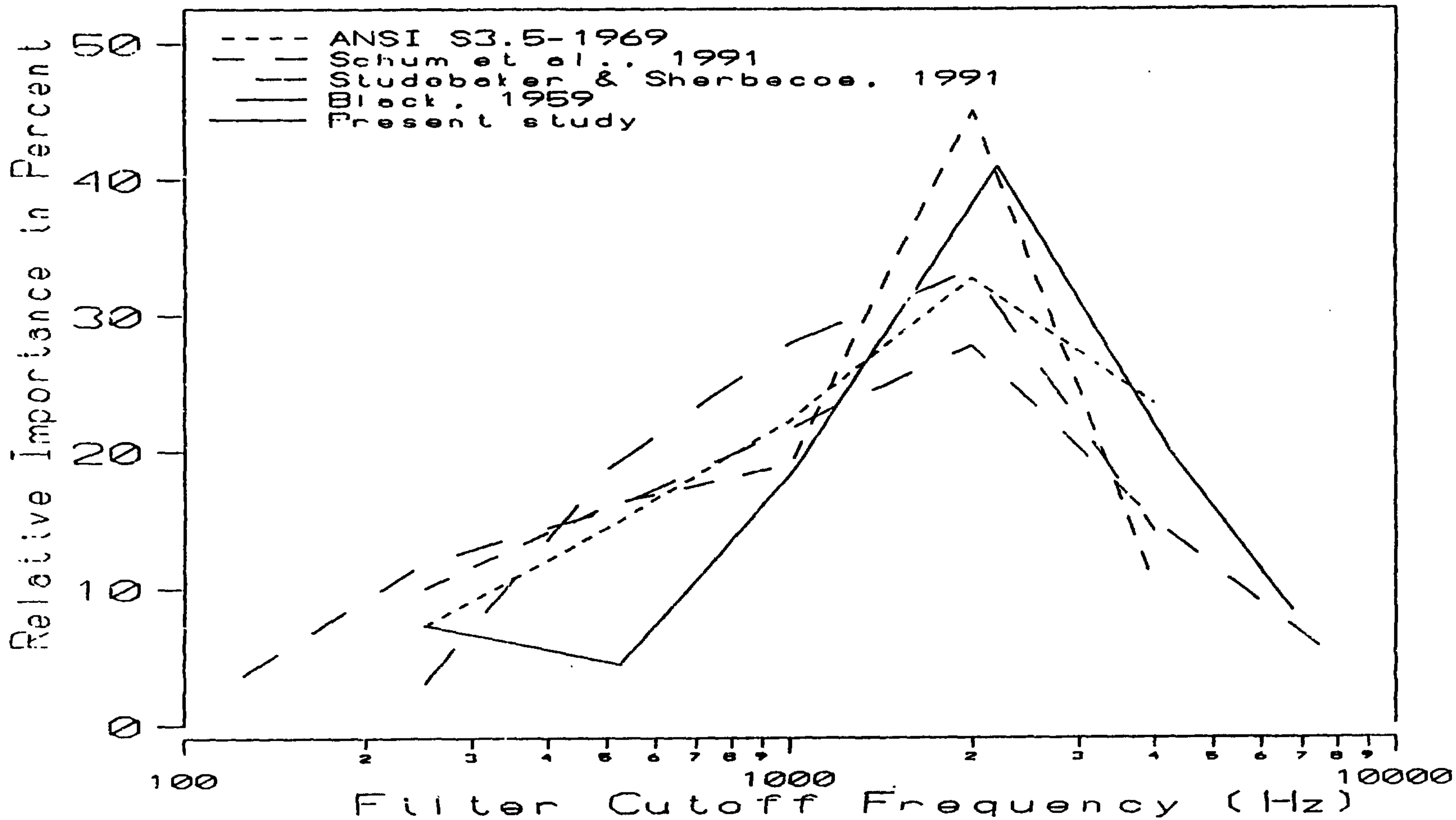


Figure 17: The Frequency Importance function for words for the present study, for NU-6 word lists (Shcum et al., 1991), for the CID W-22 word test (Studebaker and Sherbecoe, 1991) and for nonsense syllables (ANSI S3.5-1969).

Table 14: Comparison of frequency bands used to compare the frequency importance functions for the present study to past studies.

BAND #	OCTAVE BANDS	PRESENT STUDY
1	90-180	-
2	180-355	100-400
3	355-710	400-650
4	710-1400	650-1400
5	1400-2800	1400-2200
6	2800-5600	2200-5500
7	5600-11200	5500-8000

Figure 17 on page 84 reveals a similarity between the I(f) for the present study and the NU-6 word lists by Schum et al. (1991). Both peak around 2000 Hz, the area in the speech spectrum where most consonantal cues are located. The I(f) for the CID W-22 word test (Studebaker and Sherbecoe, 1991) also peaked in the same area, but similar to the I(f) for nonsense syllables, the speech information is more evenly spread throughout the speech spectrum. The fact that every study peaks around 2000 Hz emphasizes how important this area of the speech spectrum is toward the understanding of speech.

The difference between the present study and that by Studebaker and Sherbecoe (1991) illustrates the different strategies used by the subjects when the list of words is known (Studebaker and Sherbecoe, 1991) versus when the list of words is not known (present study and Schum et al., 1991) to the subjects. The effect is similar to the addition of contextual cues. The peak area of importance decreases in magnitude and spreads out to include more high and low frequencies. The reduction of the message set and the addition of context did not seem to force the subject to concentrate on a specific narrow band of frequencies. Instead, the area of concentration widens, allowing subjects to relax their focus and pick cues from a larger area of the speech spectrum.

The I(f)s for CD differs greatly from that of Studebaker et al. (1987). Figure 18 compares the I(f)s for CD and shows the octave band I(f) for sentences for the present study. The I(f)s for CD differ in both shape and emphasis. There was one major difference between the two studies. Studebaker et al. (1987) used three speakers, two male and one female versus this study that used a single male speaker. This would account for the difference in the low frequency emphasis. The disparity at high frequencies is not as easily explained. The addition of a female speaker should add high frequency emphasis. In this case, Studebaker's I(f) has a lower high frequency emphasis. There is no obvious explanation for this disparity, although the possible experimental error involved in calculating the extreme frequency band estimates in this study could account for some of the difference.

Close inspection of Figure 18 reveals that the octave band I(f)s for sentences and CD derived in this study were almost identical. This was a significant finding, especially in reference to the similarity of the sentence and CD transfer functions. The similarity of the sentence and CD transfer function and frequency importance function imply that these two types of speech can be used interchangeably, especially when using the octave band method of computing the AI. In other words, the calculation of the AI using methods similar to ANSI S3.5-1969 would be nearly identical for sentences and CD if the specific frequency importance function for either speech material were used. This observation is even more significant in the context of the tight controls employed in this study to derive the frequency importance functions. The assumptions, equipment, subjects, intsrumentation and methodology were identical.

The differences between the I(f)s for different speech material was not surprising. However, the differences between the I(f)s for the same types of speech were surprising. Recall, from chapter 2, that ANSI S3.5-1969 recommends the use of one I(f) for all types

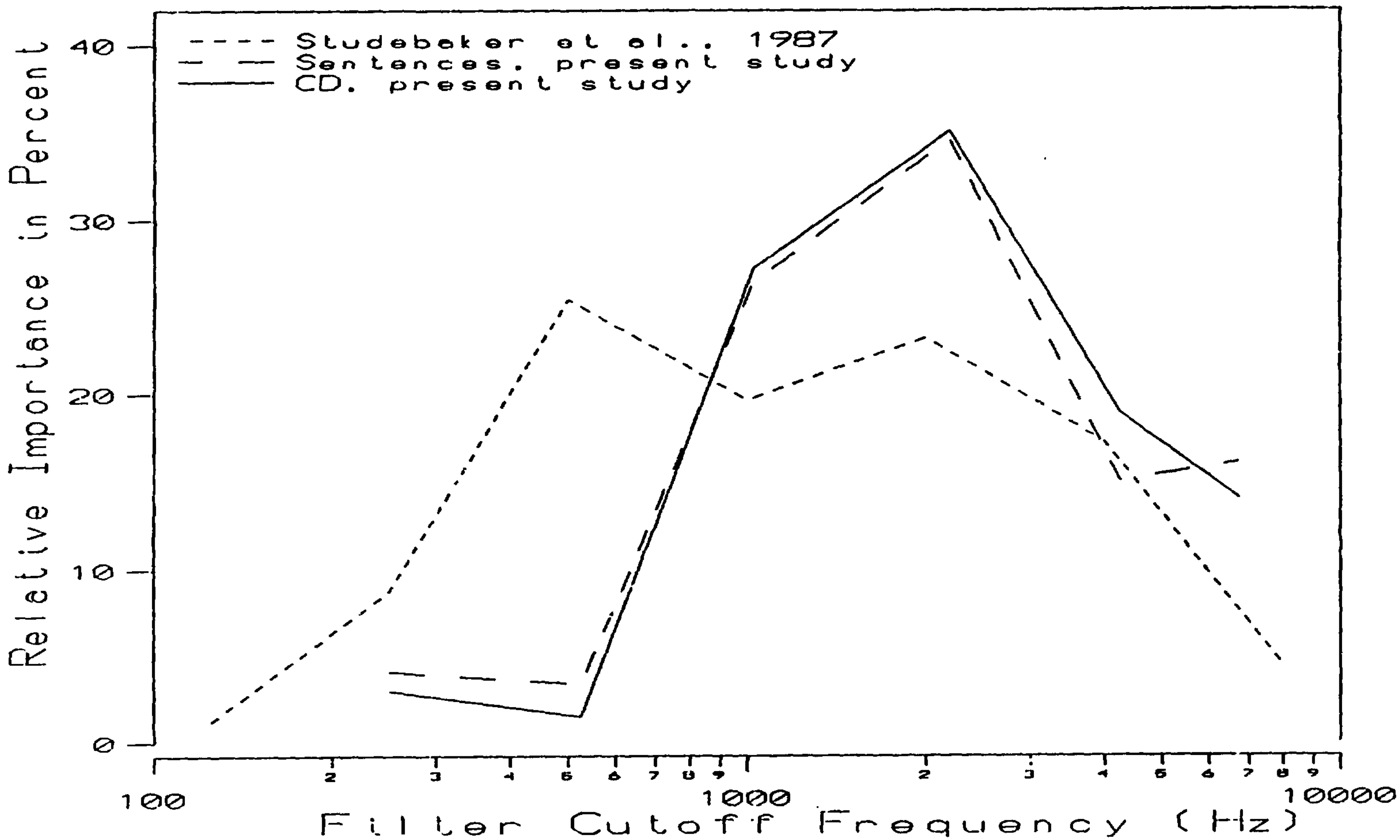


Figure 18: The frequency importance functions for CD for this study and for Studebaker et al., 1987.

of speech. A comparison of the ANSI S3.6-1969 I(f) in Figure 17 with the I(f)s for words in Figure 15 and the sentence and CD I(f)s in Figure 18 on page 87 reveals that there are major differences. Therefore the use of just one frequency importance function for all types of speech will add error into the calculation of the AI.

Future Work

The prevalence of the AI in recent research concerning both clinical (Studebaker and Sherbecoe, 1991, Berger, 1990, Bergenstoff, 1990) and non-clinical studies (Schum et al., 1991, Wilde and Humes, 1990, Williams and Michael, 1991) demonstrate the usefulness of this procedure. The applications have ranged from evaluating hearing protectors, to predicting hearing aid gain to predicting the word recognition of elderly listeners. Most applications of the AI have met with success.

One area in which surprisingly little research has been performed is to understand how the elderly listener uses acoustic information differently from the young listener. Schum et al. (1991) investigated the potential of the AI for predicting WRSs of the elderly, but interestingly, they developed a frequency importance function for their test stimuli with young normal hearing listeners. A grant proposal has been submitted to the Andrus Foundation to extend the research performed in this thesis using elderly subjects.

Another possible use of the AI is the fitting of hearing aids. There has been extensive work in the area of prescribing hearing aid gain (Berger, 1990, Pavlovic, 1988, Pavlovic, 1989). Pavlovic (1991) has recently investigated the use of the AI with a sentence estimation task to evaluate hearing aids with good results. This type of research, if taken one step further, could provide a much needed link between speech intelligibility and the sound quality of a hearing aid. The estimation judgments investigated by Pavlovic (1991)

could be followed by sound quality judgments that could aid the clinician in prescribing a hearing aid that, not only maximizes speech intelligibility but also increases the chance of the hearing aid user actually using the aid.

Finally, the data presented in this research raises some serious questions concerning the recommended procedure for calculating the AI in ANSI S3.5-1969. A detailed study could be performed evaluating some of the basic assumptions of this standard. Most notably, the assumption that one frequency importance function can be used to obtain accurate estimates of AI for different types of speech. The reseach presented in this study as well as other studies (Pavlovic, 1984, Studebaker and Sherbecoe, 1991, Schum et al., 1991, Black, 1959 and Studebaker et al., 1987) have raised some serious doubts about this assumption. Unequivocally, the results of the present study revealed that differences across speech types are not due to differences between studies, but that these differences are due to the inherent dissimilarities between speech with varying message redundancy.

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Appendix A

RAW DATA FOR WORDS

The following data is organized in blocks of low, high or all pass filters. The upper case letter (either L or H) designates a low or high pass filter respectively. Each block contains the data from each subject for the four S/N conditions used in this study. Thus, the first block that follows is designated as L400 and contains the data for the low pass filter with a cutoff frequency of 400 Hz. The four S/N ratios which are reported are -9, -3, +3 and +9 dB.

	L400 -9	L400 -3	L400 +3	L400 +9	L650 -9	L650 -3	L650 +3	L650 +9
sub1	0.0	0.0	0.0	0.0	0.0	0.0	7.1	7.1
sub2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.1
sub3	0.0	0.0	0.0	7.1	0.0	0.0	21.4	7.1
sub4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	21.4
sub5	0.0	0.0	0.0	0.0	0.0	0.0	0.0	28.4
sub6	0.0	0.0	0.0	0.0	0.0	0.0	14.3	7.1
sub7	0.0	0.0	0.0	0.0	0.0	0.0	7.1	7.1
sub8	0.0	0.0	0.0	7.1	0.0	0.0	0.0	7.1
sub9	0.0	0.0	0.0	7.1	0.0	0.0	0.0	0.0
sub10	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.1
sub11	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.1
sub12	0.0	0.0	0.0	0.0	0.0	0.0	0.0	14.3
sub13	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.1
sub14	0.0	0.0	0.0	7.1	0.0	0.0	0.0	0.0
sub15	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub16	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.1
sub17	0.0	0.0	0.0	0.0	0.0	7.1	0.0	21.4
sub18	0.0	0.0	0.0	7.1	0.0	0.0	0.0	7.1
sub19	0.0	0.0	0.0	0.0	0.0	0.0	0.0	14.3
sub20	0.0	0.0	0.0	0.0	0.0	7.1	7.1	7.1
sub21	0.0	0.0	0.0	0.0	0.0	0.0	7.1	0.0
sub22	0.0	0.0	0.0	0.0	0.0	0.0	0.0	14.3
sub23	0.0	0.0	7.1	0.0	0.0	0.0	0.0	21.4
sub24	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
avg	0.0	0.0	0.3	1.5	0.0	0.6	2.7	9.2
var	0.0	0.0	2.0	8.3	0.0	3.9	28.9	57.1

	L920 -9	L920 -3	L920 +3	L920 +9	L1400 -9	L1400 -3	L1400 +3	L1400 +9
sub1	0.0	0.0	28.6	7.1	0.0	21.4	42.9	85.7
sub2	0.0	7.1	21.4	21.4	0.0	7.1	35.7	64.3
sub3	0.0	0.0	7.1	21.4	0.0	0.0	28.6	35.7
sub4	0.0	7.1	21.4	28.6	0.0	7.1	28.6	64.3
sub5	0.0	0.0	0.0	28.4	0.0	0.0	50.0	42.9
sub6	0.0	7.1	7.1	35.7	0.0	28.6	28.6	64.3
sub7	0.0	0.0	7.1	28.6	0.0	0.0	50.0	71.4
sub8	0.0	7.1	14.3	35.6	0.0	7.1	28.6	64.6
sub9	0.0	0.0	7.1	14.3	0.0	14.3	21.4	35.7
sub10	0.0	0.0	14.3	0.0	0.0	14.3	42.9	42.9
sub11	0.0	0.0	7.1	35.7	0.0	0.0	50.0	57.1
sub12	0.0	0.0	7.1	14.3	0.0	0.0	28.6	85.7
sub13	0.0	7.1	14.3	37.5	0.0	7.1	21.4	64.3
sub14	0.0	7.1	7.1	21.4	0.0	14.3	35.7	64.3
sub15	0.0	0.0	0.0	21.4	0.0	0.0	28.6	35.6
sub16	0.0	0.0	21.4	21.4	0.0	0.0	21.4	50.0
sub17	0.0	0.0	7.1	42.9	0.0	0.0	21.4	64.6
sub18	0.0	7.1	7.1	21.4	7.1	0.0	7.1	64.3
sub19	0.0	0.0	7.1	7.1	0.0	7.1	35.5	35.7
sub20	0.0	0.0	21.4	21.4	0.0	7.1	28.4	57.1

sub21	0.0	0.0	7.1	28.4	0.0	21.4	21.4	35.4
sub22	0.0	0.0	0.0	50.0	0.0	7.1	21.4	35.7
sub23	0.0	7.1	0.0	35.7	0.0	7.1	14.3	50.0
sub24	0.0	0.0	0.0	7.1	0.0	7.1	21.3	42.9
avg	0.0	2.4	9.8	24.5	0.3	7.4	29.7	54.8
var	0.0	11.2	63.0	146.0	2.0	61.6	122.1	233.5

	L4500				all			
	-9	-3	+3	+9	-9	-3	+3	+9
sub1	7.1	50.0	100.0	92.9	21.4	42.9	85.7	85.7
sub2	0.0	50.0	71.4	100.0	14.3	28.6	92.9	100.0
sub3	7.1	50.0	64.3	100.0	28.6	50.0	85.7	100.0
sub4	35.7	78.6	71.4	85.7	7.4	57.1	92.9	100.0
sub5	0.0	42.9	85.7	78.6	7.1	64.3	85.7	100.0
sub6	14.3	71.4	78.6	92.8	7.1	64.3	78.6	85.7
sub7	0.0	64.3	85.7	100.0	14.3	78.6	100.0	92.8
sub8	14.3	64.6	100.0	92.9	0.0	64.6	92.9	78.6
sub9	0.0	35.7	64.3	92.9	21.4	42.9	85.7	92.9
sub10	7.1	28.6	64.3	92.9	7.1	64.3	78.6	92.9
sub11	0.0	42.9	85.7	78.6	0.0	42.9	71.4	100.0
sub12	0.0	42.9	78.6	92.9	0.0	28.6	85.7	92.9
sub13	14.3	64.3	92.8	85.7	14.3	42.9	92.8	78.6
sub14	7.1	57.1	92.9	85.7	28.4	35.7	92.9	92.9
sub15	0.0	28.6	42.9	71.4	21.4	14.3	71.4	100.0
sub16	0.0	14.3	78.6	100.0	[illegible]	57.1	78.6	78.6
sub17	14.3	50.0	78.6	85.6	[illegible]	64.6	71.4	92.9
sub18	7.1	42.9	57.1	85.7	[illegible]	64.3	78.6	100.0
sub19	7.1	28.4	42.9	100.0	[illegible]	64.3	78.6	64.1
sub20	0.0	28.4	85.7	71.1	0.0	42.9	78.6	100.0
sub21	7.1	57.1	78.4	85.7	0.0	50.0	85.7	85.7
sub22	0.0	42.9	72.4	85.7	7.1	35.7	78.5	78.6
sub23	0.0	35.7	50.0	92.9	7.1	57.1	85.7	92.9
sub24	7.1	42.9	42.9	85.7	21.3	42.9	71.4	85.7
avg	6.2	46.4	73.6	89.0	11.9	50.0	83.3	90.5
var	65.1	230.4	282.1	68.5	74.8	221.7	62.5	88.2

	H920				H1400			
	-9	-3	+3	+9	-9	-3	+3	+9
sub1	14.3	61.5	85.7	100.0	14.3	57.1	71.4	92.9
sub2	0.0	42.9	78.6	78.6	7.1	57.1	71.4	100.0
sub3	28.6	57.1	71.4	92.9	7.1	21.4	57.1	71.4
sub4	42.9	42.9	50.0	78.6	7.1	28.6	50.0	85.7
sub5	0.0	42.9	71.4	78.6	7.1	35.7	64.3	85.7
sub6	35.7	42.9	57.1	78.6	0.0	35.7	78.6	64.3
sub7	21.4	21.4	71.4	85.7	7.1	57.1	71.4	64.3
sub8	35.6	35.6	57.1	71.4	7.1	50.0	57.1	85.7
sub9	7.1	50.0	71.4	78.6	0.0	57.1	57.1	64.3
sub10	14.3	50.0	71.4	92.9	21.4	64.3	71.4	57.1
sub11	14.3	21.3	71.4	78.6	21.4	42.9	57.1	64.3
sub12	0.0	42.8	64.3	71.4	0.0	28.6	42.9	85.7

sub13	35.7	42.9	78.6	92.8	7.1	50.0	57.1	92.8
sub14	14.3	35.6	64.3	85.7	7.1	42.9	64.3	92.9
sub15	14.3	7.1	35.7	71.4	0.0	21.4	78.6	71.4
sub16	7.1	35.7	57.1	71.4	0.0	57.1	64.3	78.6
sub17	14.3	42.9	64.6	92.9	0.0	35.7	57.1	85.6
sub18	0.0	50.0	71.4	85.7	0.0	14.3	71.4	92.9
sub19	14.3	28.5	71.4	64.1	14.3	28.5	57.1	50.0
sub20	7.1	64.1	64.1	92.9	14.3	14.3	50.0	92.9
sub21	7.1	35.7	50.0	78.4	7.1	28.4	71.1	85.7
sub22	21.4	42.9	72.6	64.6	0.0	50.0	35.7	78.6
sub23	28.4	21.4	78.6	78.6	0.0	57.1	42.9	85.7
sub24	7.1	50.0	71.4	78.6	0.0	28.4	64.3	78.6
avg	16.1	40.3	66.7	81.0	6.2	40.2	61.0	79.5
var	154.0	172.1	118.2	87.9	43.8	224.6	122.9	167.2

	H2200				H3000			
	-9	-3	+3	+9	-9	-3	+3	+9
sub1	0.0	21.4	58.3	64.3	0.0	21.4	0.0	35.7
sub2	7.1	28.6	42.9	50.0	0.0	7.1	35.7	28.6
sub3	7.1	21.4	35.7	57.1	0.0	7.1	14.3	42.9
sub4	0.0	21.4	14.3	38.4	7.1	21.4	14.3	28.6
sub5	7.1	7.1	14.3	50.0	0.0	14.3	7.1	28.4
sub6	7.1	14.3	14.3	42.9	0.0	7.1	21.4	28.6
sub7	21.4	28.6	42.8	50.0	0.0	7.1	14.3	7.1
sub8	7.1	14.3	7.1	42.9	0.0	21.4	21.4	14.3
sub9	14.3	14.3	42.9	35.7	0.0	14.3	21.4	21.4
sub10	0.0	14.3	35.7	50.0	0.0	14.3	28.6	35.7
sub11	7.1	14.3	21.4	28.6	7.1	14.3	7.1	21.4
sub12	0.0	14.3	42.9	42.8	0.0	7.1	0.0	7.1
sub13	14.3	21.4	14.3	35.7	0.0	21.4	14.3	21.4
sub14	0.0	21.4	50.0	71.4	0.0	7.1	28.4	50.0
sub15	0.0	7.1	57.1	35.7	0.0	0.0	7.1	7.1
sub16	7.1	28.6	14.3	28.6	0.0	14.3	28.7	21.4
sub17	7.1	14.3	14.3	57.1	0.0	0.0	7.1	28.6
sub18	0.0	14.3	35.7	64.3	0.0	0.0	7.1	50.0
sub19	0.0	14.3	42.9	35.6	7.1	0.0	14.3	28.4
sub20	0.0	21.3	35.6	64.1	0.0	7.1	0.0	28.4
sub21	0.0	14.3	28.4	64.3	0.0	14.3	0.0	21.4
sub22	0.0	0.0	28.4	35.7	7.1	14.3	7.1	21.4
sub23	0.0	28.4	64.6	71.4	0.0	7.1	28.4	14.3
sub24	0.0	7.1	35.7	64.3	0.0	14.3	28.4	7.1
avg	4.5	17.0	33.1	49.2	1.2	10.7	14.9	25.0
var	33.1	54.4	249.1	176.9	7.0	46.8	114.1	144.8

	H5500			
	-9	-3	+3	+9
sub1	0.0	0.0	7.1	0.0
sub2	0.0	0.0	7.1	14.3
sub3	0.0	7.1	14.3	0.0
sub4	0.0	0.0	0.0	0.0

sub5	0.0	0.0	7.1	0.0
sub6	0.0	0.0	7.1	14.3
sub7	7.1	7.1	0.0	7.1
sub8	0.0	0.0	0.0	0.0
sub9	0.0	0.0	0.0	0.0
sub10	0.0	0.0	0.0	0.0
sub11	7.1	0.0	0.0	0.0
sub12	0.0	0.0	0.0	0.0
sub13	0.0	0.0	0.0	0.0
sub14	0.0	0.0	7.1	7.1
sub15	0.0	0.0	0.0	0.0
sub16	0.0	0.0	7.1	7.1
sub17	0.0	0.0	7.1	0.0
sub18	0.0	7.1	7.1	0.0
sub19	0.0	0.0	0.0	0.0
sub20	0.0	0.0	7.1	0.0
sub21	0.0	0.0	7.1	0.0
sub22	0.0	0.0	7.1	0.0
sub23	0.0	7.1	7.1	21.4
sub24	0.0	14.3	0.0	0.0
avg	0.6	1.8	4.1	3.0
var	3.9	13.8	16.5	33.6

Appendix B

RAW DATA FOR SENTENCES

The following data is organized in blocks of low. high or all pass filters. The upper case letter (either L or H) designates a low or high pass filter respectively. Each block contains the data from each subject for the four S/N conditions used in this study. Thus, the first block that follows is designated as L400 and contains the data for the low pass filter with a cutoff frequency of 400 Hz. The four S/N ratios which are reported are -12, -8, -4 and 0 dB.

Raw data for sentences

	L400				L650			
	-12	-8	-4	0	-12	-8	-4	0
sub1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	37.5
sub2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	15.0
sub4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub5	0.0	0.0	0.0	7.5	0.0	0.0	0.0	7.5
sub6	0.0	0.0	0.0	7.5	0.0	0.0	0.0	7.5
sub7	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub8	0.0	0.0	0.0	15.0	0.0	0.0	7.5	7.5
sub9	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub10	0.0	0.0	0.0	7.5	0.0	0.0	0.0	0.0
sub11	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub12	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub13	0.0	0.0	0.0	22.5	0.0	0.0	30.0	15.0
sub14	0.0	0.0	0.0	0.0	0.0	0.0	0.0	15.0
sub15	0.0	0.0	0.0	0.0	0.0	0.0	22.5	7.5
sub16	0.0	0.0	0.0	7.5	0.0	0.0	0.0	7.5
sub17	0.0	0.0	0.0	0.0	0.0	0.0	7.5	7.5
sub18	0.0	0.0	0.0	0.0	0.0	0.0	0.0	15.0
sub19	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub20	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub21	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub22	0.0	0.0	0.0	7.5	0.0	0.0	0.0	7.5
sub23	0.0	0.0	0.0	0.0	0.0	0.0	22.5	15.0
sub24	0.0	0.0	0.0	7.5	0.0	0.0	7.5	7.5
avg	0.0	0.0	0.0	3.4	0.0	0.0	4.1	8.4
var	0.0	0.0	0.0	5.7	0.0	0.0	8.4	7.9

	L920				L1400			
	-12	-8	-4	0	-12	-8	-4	0
sub1	0.0	0.0	7.5	67.5	15.0	30.0	45.0	100.0
sub2	0.0	0.0	0.0	22.5	0.0	0.0	52.5	45.0
sub3	0.0	0.0	0.0	45.0	0.0	0.0	30.0	75.0
sub4	0.0	0.0	0.0	67.5	15.0	45.0	60.0	100.0
sub5	0.0	0.0	7.5	30.0	0.0	0.0	45.0	82.5
sub6	0.0	0.0	0.0	22.5	0.0	0.0	60.0	90.0
sub7	0.0	0.0	22.5	15.0	0.0	15.0	37.5	82.5
sub8	0.0	0.0	0.0	37.5	7.5	30.0	45.0	67.5
sub9	0.0	0.0	0.0	7.5	15.0	15.0	60.0	75.0
sub10	0.0	0.0	0.0	22.5	0.0	7.5	22.5	100.0
sub11	0.0	0.0	0.0	22.5	0.0	7.5	67.5	90.0
sub12	0.0	0.0	0.0	0.0	0.0	0.0	52.5	75.0
sub13	0.0	0.0	37.5	37.5	15.0	22.5	60.0	82.5
sub14	0.0	0.0	30.0	37.5	7.5	22.5	60.0	67.5
sub15	0.0	0.0	7.5	60.0	0.0	7.5	37.5	82.5
sub16	0.0	0.0	45.0	75.0	7.5	7.5	67.5	67.5
sub17	0.0	7.5	7.5	30.0	0.0	7.5	60.0	90.0

sub18	0.0	0.0	37.5	52.5	0.0	37.5	45.0	97.5
sub19	0.0	0.0	15.0	30.0	15.0	22.5	30.0	97.5
sub20	0.0	0.0	0.0	30.0	0.0	7.5	82.5	75.0
sub21	0.0	7.5	0.0	7.5	0.0	0.0	52.5	60.0
sub22	0.0	7.5	22.5	60.0	7.5	0.0	45.0	82.5
sub23	0.0	0.0	0.0	15.0	7.5	15.0	45.0	82.5
sub24	0.0	0.0	7.5	7.5	7.5	22.5	45.0	75.0
avg	0.0	0.9	10.3	33.4	5.0	13.4	50.3	80.9
var	0.0	2.5	14.0	20.8	6.0	12.8	13.4	13.5

	L4500				all			
	-12	-8	-4	0	-12	-8	-4	0
sub1	30.0	82.5	75.0	100.0	30.0	90.0	100.0	100.0
sub2	7.5	22.5	90.0	100.0	0.0	75.0	100.0	100.0
sub3	7.5	37.5	100.0	97.5	7.5	67.5	100.0	97.5
sub4	67.5	75.0	100.0	100.0	60.0	97.5	100.0	100.0
sub5	7.5	15.0	90.0	90.0	0.0	30.0	90.0	100.0
sub6	7.5	37.5	52.5	97.5	7.5	52.5	97.5	97.5
sub7	15.0	67.5	67.5	100.0	15.0	37.5	82.5	100.0
sub8	7.5	60.0	97.5	100.0	37.5	75.0	90.0	100.0
sub9	0.0	67.5	90.0	97.5	37.5	67.5	100.0	100.0
sub10	15.0	75.0	97.5	100.0	30.0	60.0	100.0	100.0
sub11	22.5	60.0	97.5	100.0	60.0	82.5	90.0	100.0
sub12	0.0	15.0	97.5	100.0	52.5	52.5	100.0	100.0
sub13	7.5	75.0	90.0	100.0	45.0	67.5	90.0	100.0
sub14	7.5	37.5	100.0	100.0	22.5	52.5	97.5	100.0
sub15	37.5	37.5	97.5	100.0	30.0	37.5	100.0	100.0
sub16	0.0	52.5	100.0	100.0	52.5	82.5	100.0	100.0
sub17	30.0	22.5	75.0	100.0	15.0	82.5	82.5	100.0
sub18	7.5	37.5	97.5	100.0	7.5	82.5	100.0	100.0
sub19	15.0	52.5	90.0	100.0	22.5	90.0	97.5	100.0
sub20	22.5	97.5	100.0	100.0	30.0	15.0	100.0	100.0
sub21	7.5	15.0	97.5	100.0	45.0	75.0	82.5	97.5
sub22	30.0	75.0	100.0	100.0	7.5	75.0	100.0	100.0
sub23	30.0	7.5	100.0	97.5	7.5	52.5	100.0	100.0
sub24	15.0	37.5	90.0	100.0	7.5	7.5	100.0	100.0
avg	16.6	48.4	91.4	99.2	26.3	62.8	95.8	99.7
var	15.0	24.6	11.9	2.1	18.8	23.3	6.2	0.8

	H920				H1400			
	-12	-8	-4	0	-12	-8	-4	0
sub1	45.0	37.5	82.5	100.0	67.5	52.5	82.5	100.0
sub2	0.0	45.0	100.0	100.0	0.0	15.0	100.0	100.0
sub3	15.0	22.5	75.0	100.0	15.0	52.5	97.5	97.5
sub4	52.5	75.0	75.0	100.0	0.0	90.0	60.0	100.0
sub5	15.0	15.0	90.0	100.0	7.5	30.0	97.5	100.0
sub6	7.5	75.0	82.5	82.5	7.5	37.5	82.5	82.5
sub7	45.0	45.0	97.5	97.5	0.0	60.0	90.0	100.0
sub8	30.0	75.0	82.5	100.0	60.0	60.0	75.0	100.0
sub9	15.0	67.5	97.5	100.0	0.0	60.0	90.0	97.5

sub10	7.5	45.0	97.5	100.0	15.0	22.5	60.0	90.0
sub11	37.5	60.0	97.5	100.0	30.0	60.0	97.5	100.0
sub12	22.5	0.0	97.5	97.5	0.0	0.0	90.0	97.5
sub13	7.5	67.5	82.5	100.0	45.0	67.5	97.5	97.5
sub14	22.5	30.0	90.0	100.0	15.0	37.5	82.5	75.0
sub15	30.0	52.5	82.5	100.0	7.5	37.5	82.5	100.0
sub16	37.5	52.5	90.0	100.0	52.5	82.5	97.5	75.0
sub17	22.5	30.0	97.5	100.0	7.5	75.0	75.0	100.0
sub18	15.0	30.0	97.5	100.0	7.5	67.5	97.5	100.0
sub19	37.5	52.5	100.0	100.0	15.0	75.0	97.5	90.0
sub20	15.0	52.5	100.0	100.0	22.5	67.5	90.0	97.5
sub21	0.0	52.5	100.0	100.0	7.5	22.5	75.0	100.0
sub22	52.5	97.5	90.0	97.5	15.0	37.5	90.0	100.0
sub23	45.0	30.0	82.5	97.5	7.5	0.0	75.0	100.0
sub24	52.5	82.5	100.0	100.0	30.0	52.5	100.0	100.0
avg	26.3	49.7	91.1	98.9	18.1	48.4	86.8	95.8
var	16.6	22.6	8.3	3.5	19.2	24.1	11.7	7.6

	H2200				H3000			
	-12	-8	-4	0	-12	-8	-4	0
sub1	7.5	22.5	67.5	82.5	0.0	30.0	52.5	60.0
sub2	7.5	45.0	22.5	52.5	0.0	7.5	0.0	7.5
sub3	7.5	7.5	67.5	90.0	15.0	7.5	60.0	52.5
sub4	15.0	30.0	82.5	75.0	0.0	30.0	45.0	0.0
sub5	7.5	7.5	22.5	75.0	0.0	0.0	37.5	22.5
sub6	0.0	7.5	45.0	22.5	0.0	7.5	15.0	30.0
sub7	0.0	7.5	0.0	82.5	0.0	0.0	22.5	37.5
sub8	7.5	7.5	37.5	52.5	15.0	15.0	60.0	67.5
sub9	0.0	7.5	45.0	82.5	0.0	0.0	22.5	75.0
sub10	0.0	22.5	75.0	60.0	0.0	7.5	30.0	22.5
sub11	0.0	15.0	82.5	97.5	0.0	7.5	15.0	7.5
sub12	0.0	0.0	22.5	22.5	0.0	0.0	7.5	37.5
sub13	15.0	15.0	37.5	90.0	22.5	15.0	45.0	37.5
sub14	15.0	22.5	52.5	82.5	0.0	7.5	7.5	45.0
sub15	15.0	7.5	15.0	90.0	22.5	0.0	45.0	60.0
sub16	7.5	7.5	45.0	97.5	0.0	0.0	30.0	67.5
sub17	7.5	30.0	37.5	67.5	0.0	15.0	30.0	52.5
sub18	0.0	15.0	45.0	82.5	0.0	7.5	7.5	30.0
sub19	0.0	30.0	37.5	90.0	15.0	15.0	22.5	37.5
sub20	15.0	7.5	75.0	75.0	0.0	0.0	0.0	60.0
sub21	7.5	15.0	30.0	37.5	0.0	0.0	0.0	22.5
sub22	7.5	67.5	45.0	60.0	0.0	7.5	7.5	52.5
sub23	7.5	67.5	37.5	60.0	0.0	0.0	0.0	22.5
sub24	7.5	15.0	67.5	82.5	7.5	15.0	45.0	60.0
avg	6.6	20.0	45.6	71.3	4.1	8.1	25.3	40.3
var	5.4	17.5	21.5	21.0	7.5	8.6	19.3	20.4

	H5500			
	-12	-8	-4	0
sub1	7.5	7.5	15.0	52.5

sub2	0.0	0.0	0.0	0.0
sub3	0.0	7.5	7.5	15.0
sub4	0.0	0.0	7.5	45.0
sub5	0.0	0.0	0.0	7.5
sub6	7.5	7.5	0.0	7.5
sub7	0.0	0.0	0.0	0.0
sub8	0.0	22.5	60.0	22.5
sub9	0.0	0.0	0.0	7.1
sub10	0.0	0.0	7.5	0.0
sub11	0.0	0.0	0.0	0.0
sub12	0.0	0.0	0.0	0.0
sub13	0.0	7.5	22.5	37.5
sub14	0.0	7.5	15.0	7.5
sub15	22.5	30.0	0.0	30.0
sub16	0.0	15.0	15.0	7.5
sub17	0.0	0.0	0.0	15.0
sub18	0.0	0.0	15.0	15.0
sub19	0.0	7.5	15.0	15.0
sub20	0.0	0.0	0.0	0.0
sub21	7.5	0.0	7.5	0.0
sub22	7.5	0.0	7.5	15.0
sub23	0.0	0.0	30.0	7.5
sub24	0.0	7.5	7.5	30.0
avg	2.2	5.0	9.7	14.0
var	5.1	7.7	13.3	14.7

Appendix C

RAW DATA FOR CD

The following data is organized in blocks of low, high or all pass filters. The upper case letter (either L or H) designates a low or high pass filter respectively. Each block contains the data from each subject for the four S/N conditions used in this study. Thus, the first block that follows is designated as L400 and contains the data for the low pass filter with a cutoff frequency of 400 Hz. The four S/N ratios which are reported are -12, -9, -6 and -3 dB.

Raw data for CD

	L400				L650			
	-12	-9	-6	-3	-12	-9	-6	-3
sub1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub5	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub6	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub7	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub8	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub9	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub10	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub11	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub12	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub13	0.0	0.0	0.0	7.5	0.0	0.0	0.0	7.5
sub14	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub15	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub16	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub17	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub18	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub19	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
sub20	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub21	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub22	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub23	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
sub24	0.0	0.0	0.0	0.0	0.0	0.0	0.0	7.5
avg	0.0	0.0	0.0	0.3	0.0	0.0	0.0	3.4
var	0.0	0.0	0.0	1.5	0.0	0.0	0.0	3.7

	L920				L1400			
	-12	-9	-6	-3	-12	-9	-6	-3
sub1	0.0	0.0	7.5	22.5	0.0	0.0	30.0	90.0
sub2	0.0	0.0	7.5	7.5	0.0	0.0	7.5	30.0
sub3	0.0	0.0	15.0	7.5	7.5	0.0	22.5	15.0
sub4	0.0	0.0	7.5	22.5	0.0	0.0	37.5	90.0
sub5	0.0	0.0	7.5	7.5	7.5	0.0	0.0	22.5
sub6	0.0	0.0	0.0	50.0	0.0	0.0	0.0	75.0
sub7	0.0	0.0	7.5	7.5	0.0	0.0	7.5	75.0
sub8	0.0	0.0	0.0	15.0	0.0	0.0	0.0	60.0
sub9	0.0	0.0	7.5	22.5	0.0	0.0	7.5	30.0
sub10	0.0	0.0	0.0	15.0	0.0	0.0	0.0	22.5
sub11	0.0	0.0	0.0	0.0	0.0	0.0	7.5	15.0
sub12	0.0	0.0	0.0	0.0	0.0	0.0	0.0	22.5
sub13	0.0	7.5	45.0	37.5	15.0	0.0	22.5	75.0
sub14	0.0	0.0	15.0	7.5	0.0	0.0	0.0	75.0
sub15	0.0	30.0	15.0	37.5	0.0	0.0	30.0	22.5
sub16	0.0	0.0	0.0	22.5	0.0	0.0	52.5	60.0
sub17	0.0	0.0	0.0	7.5	0.0	0.0	7.5	7.5

sub18	0.0	0.0	15.0	15.0	0.0	0.0	7.5	45.0
sub19	0.0	7.5	7.5	7.5	0.0	0.0	7.5	7.5
sub20	0.0	0.0	0.0	7.5	0.0	0.0	15.0	60.0
sub21	0.0	7.5	22.5	45.0	0.0	0.0	0.0	60.0
sub22	0.0	0.0	7.5	60.0	0.0	0.0	7.5	45.0
sub23	0.0	0.0	0.0	7.5	0.0	7.5	7.5	75.0
sub24	0.0	0.0	7.5	0.0	7.5	15.0	7.5	75.0
avg	0.0	2.2	8.1	18.0	1.6	0.9	11.9	48.1
var	0.0	6.3	9.9	16.3	3.7	3.3	13.5	26.8

	L4500				all			
	-12	-9	-6	-3	-12	-9	-6	-3
sub1	7.5	30.0	52.5	100.0	37.5	67.5	100.0	100.0
sub2	7.5	0.0	45.0	97.5	0.0	7.5	97.5	100.0
sub3	30.0	15.0	97.5	97.5	15.0	45.0	97.5	100.0
sub4	7.5	82.5	82.5	100.0	22.5	45.0	100.0	100.0
sub5	15.0	0.0	60.0	100.0	7.5	30.0	75.0	97.5
sub6	7.5	30.0	52.5	90.0	15.0	7.5	82.5	90.0
sub7	0.0	30.0	82.5	100.0	7.5	30.0	97.5	97.5
sub8	15.0	30.0	90.0	60.0	7.5	22.5	97.5	75.0
sub9	0.0	30.0	45.0	100.0	15.0	15.0	97.5	100.0
sub10	0.0	7.5	82.5	100.0	15.0	67.5	90.0	100.0
sub11	52.5	0.0	100.0	100.0	22.5	60.0	90.0	97.5
sub12	0.0	52.5	90.0	100.0	0.0	37.5	90.0	97.5
sub13	15.0	37.5	45.0	97.5	22.5	30.0	82.5	90.0
sub14	22.5	15.0	67.5	100.0	30.0	67.5	67.5	97.5
sub15	7.5	45.0	52.5	82.5	7.5	75.0	67.5	97.5
sub16	7.5	37.5	100.0	97.5	30.0	60.0	100.0	97.5
sub17	0.0	7.5	52.5	75.0	7.5	22.5	90.0	90.0
sub18	7.5	0.0	97.5	97.5	22.5	75.0	90.0	100.0
sub19	30.0	15.0	75.0	90.0	15.0	45.0	75.0	90.0
sub20	37.5	67.5	90.0	100.0	75.0	67.5	90.0	100.0
sub21	7.5	22.5	82.5	97.5	7.5	52.5	97.5	100.0
sub22	0.0	52.5	100.0	100.0	45.0	60.0	100.0	100.0
sub23	7.5	22.5	100.0	100.0	37.5	15.0	97.5	97.5
sub24	7.5	15.0	90.0	100.0	45.0	45.0	52.5	100.0
avg	12.2	26.9	76.4	95.1	21.3	43.8	88.5	96.5
var	13.1	21.3	20.0	9.6	17.0	21.3	12.5	5.7

	H920				H1400			
	-12	-9	-6	-3	-12	-9	-6	-3
sub1	37.5	37.5	100.0	100.0	15.0	15.0	67.5	90.0
sub2	7.5	7.5	97.5	75.0	0.0	0.0	67.5	82.5
sub3	30.0	37.5	52.5	97.5	22.5	37.5	22.5	82.5
sub4	30.0	75.0	100.0	100.0	22.5	67.5	97.5	90.0
sub5	7.5	60.0	97.5	60.0	15.0	7.5	75.0	90.0
sub6	7.5	7.5	90.0	75.0	0.0	7.5	37.5	90.0
sub7	15.0	30.0	82.5	97.5	7.5	30.0	97.5	90.0
sub8	37.5	82.5	90.0	75.0	7.5	15.0	60.0	90.0
sub9	7.5	7.5	82.5	97.5	0.0	0.0	75.0	82.5

sub10	0.0	45.0	100.0	100.0	0.0	15.0	90.0	97.5
sub11	7.5	30.0	90.0	97.5	15.0	7.5	45.0	97.5
sub12	7.5	52.5	97.5	100.0	0.0	0.0	30.0	82.5
sub13	15.0	60.0	100.0	90.0	37.5	30.0	82.5	75.0
sub14	22.5	22.5	97.5	75.0	60.0	45.0	30.0	90.0
sub15	15.0	22.5	82.5	97.5	15.0	45.0	30.0	67.5
sub16	0.0	37.5	100.0	100.0	7.5	82.5	97.5	100.0
sub17	0.0	0.0	82.5	100.0	0.0	0.0	45.0	90.0
sub18	7.5	52.5	100.0	100.0	15.0	60.0	75.0	90.0
sub19	7.5	22.5	90.0	97.5	15.0	15.0	75.0	97.5
sub20	52.5	15.0	97.5	97.5	45.0	7.5	52.5	97.5
sub21	30.0	45.0	90.0	100.0	7.5	22.5	52.5	97.5
sub22	7.5	30.0	97.5	100.0	7.5	7.5	45.0	97.5
sub23	15.0	22.5	97.5	90.0	0.0	7.5	75.0	82.5
sub24	7.5	22.5	97.5	100.0	7.5	15.0	37.5	90.0
avg	15.6	34.4	92.2	92.6	13.4	22.5	60.9	89.2
var	13.5	21.1	10.3	11.2	15.0	22.4	23.0	7.7

	H2200				H3000			
	-12	-9	-6	-3	-12	-9	-6	-3
sub1	7.5	15.0	45.0	52.5	0.0	0.0	22.5	15.0
sub2	0.0	7.5	7.5	22.5	0.0	0.0	0.0	0.0
sub3	7.5	22.5	30.0	52.5	0.0	0.0	7.5	7.5
sub4	22.5	30.0	82.5	37.5	0.0	0.0	22.5	52.5
sub5	0.0	15.0	30.0	37.5	0.0	7.5	7.5	7.5
sub6	7.5	7.5	22.5	37.5	7.5	15.0	15.0	22.5
sub7	7.5	7.5	45.0	75.0	0.0	0.0	0.0	22.5
sub8	15.0	15.0	60.0	60.0	0.0	0.0	15.0	7.5
sub9	0.0	22.5	22.5	30.0	0.0	0.0	0.0	37.5
sub10	0.0	0.0	22.5	45.0	0.0	0.0	0.0	15.0
sub11	7.5	0.0	22.5	67.5	0.0	0.0	7.5	15.0
sub12	0.0	0.0	45.0	7.5	0.0	0.0	0.0	0.0
sub13	15.0	45.0	67.5	67.5	0.0	22.5	30.0	67.5
sub14	22.5	7.5	30.0	60.0	0.0	7.5	0.0	22.5
sub15	7.5	7.5	37.5	75.0	15.0	0.0	0.0	22.5
sub16	0.0	0.0	67.5	97.5	0.0	0.0	30.0	22.5
sub17	0.0	0.0	45.0	52.5	0.0	0.0	0.0	0.0
sub18	7.5	7.5	22.5	67.5	0.0	0.0	7.5	7.5
sub19	0.0	15.0	37.5	60.0	7.5	7.5	15.0	22.5
sub20	7.5	0.0	67.5	75.0	0.0	7.5	7.5	7.5
sub21	15.0	0.0	52.5	52.5	7.5	0.0	0.0	7.5
sub22	0.0	15.0	7.5	67.5	0.0	0.0	0.0	22.5
sub23	15.0	0.0	7.5	90.0	0.0	0.0	22.5	37.5
sub24	15.0	15.0	75.0	45.0	0.0	7.5	15.0	37.5
avg	7.5	10.6	39.7	55.6	1.6	3.1	9.4	20.0
var	7.2	11.0	21.4	20.4	3.7	5.7	10.0	16.4

	H5500			
	-12	-9	-6	-3
sub1	0.0	0.0	0.0	15.0

sub2	0.0	0.0	0.0	0.0
sub3	0.0	0.0	0.0	0.0
sub4	0.0	0.0	7.5	7.5
sub5	0.0	0.0	0.0	0.0
sub6	7.5	0.0	7.5	0.0
sub7	0.0	0.0	0.0	7.5
sub8	7.5	0.0	0.0	0.0
sub9	0.0	0.0	0.0	0.0
sub10	0.0	0.0	0.0	0.0
sub11	0.0	0.0	0.0	0.0
sub12	0.0	0.0	0.0	0.0
sub13	0.0	0.0	7.5	15.0
sub14	0.0	0.0	0.0	15.0
sub15	7.5	0.0	7.5	7.5
sub16	0.0	0.0	7.5	15.0
sub17	0.0	0.0	7.5	7.5
sub18	0.0	0.0	0.0	7.5
sub19	0.0	0.0	7.5	0.0
sub20	0.0	0.0	0.0	7.5
sub21	0.0	0.0	7.5	37.5
sub22	0.0	0.0	0.0	0.0
sub23	0.0	0.0	0.0	7.5
sub24	0.0	0.0	0.0	15.0
avg	0.9	0.0	2.5	6.9
var	2.5	0.0	3.5	8.6

Appendix D

COMPUTER PROGRAMS

```
    {         DSP-16 to Turbo Pascal 3.0 Interface             }
    {    Program reads an existing subject file which is       }
    {         specified by user and then loads files           }
    {         and filters them as spec'd by the file           }
    {                 R.A.D. 10-15-89                          }
program subject;

const
   DefSr: Integer = 69;
   DefFile: string[14] ='a:filter.hex'#0;
   DspAddr: Integer = $33C;
   DspPage: Integer = $E000;
   DspInt: Integer = 0;
   IntNo = $63;
   ende: integer = 128;       {543744 words}
   blkln : integer = 1024;
   tempram: string[14] = 'c:tsamp';
   cphrase: string[12] = 'd:cphrase';
type
   StrParm = string[14];
   ArrParm = array[0..1] of Integer;
   Parms = record
      Fun: Integer;
      RetCode: Integer;
      P2: Integer;
      P3: Integer;
      P4: Integer;
      P5: Integer;
      P6: ^ArrParm;
      P7: ^ArrParm;
      P8: ^ArrParm;
      P9: ^Integer;
      P10: ^StrParm;
      P11: ^StrParm;
   end;
   Result = record
            AX,BX,CX,DX,BP,SI,DI,DS,ES,Flags: Integer;
            end;
var
   pause: boolean;
   order: text;
   Buffer: array[1..2048] of byte;
   Zbuff: array[1..2048] of byte;
   ErrFlg: Boolean;
   Dcall: Parms;
   Regs: Result;
   IntSeg: Integer absolute $0000:$018E;
   IntOfs: Integer absolute $0000:$018C;
   Samples,tram,carry: file;
   len: string[1];
```

```
   fftype: string[2];
   init: string[4];
   sampname,sampname2:  string[7];
   ordername: string[20];
   Histop,Lostop,L2,SNR,SNR2,loadcp,l1,
   block,sync: integer;
   High,Low,flag,I,J,L,lower,templ,temph,
   ftype,code,hi,lo,dec : integer;
procedure Dsp16;
begin
   Regs.ES := Seg(Dcall);
   Regs.SI := Ofs(Dcall);
   Intr(IntNo, Regs);
   end;

procedure WaitKP;
begin
   Writeln('');
   Writeln('Press any key to continue...');
   repeat until KeyPressed;
end;

procedure Derror(Frame: Parms);
begin
   ErrFlg := True;
Writeln('');
Writeln('Error ', Frame.RetCode,'
in DSP-16 call ', Frame.Fun);
Writeln('#1:',Frame.P2,'  #2:',Frame.P3,'
  #3:',Frame.P4,'  #4:',Frame.P5,
            '   #5:',seg(Frame.P6),'  #6:',ofs(Frame.p6));
   Sound(500);
   Delay(200);
   NoSound;
   WaitKP;
end;
{This procedure increments the low order
bit when it is below 32768, neccessary since Tpascal version
3 does
not have 16 bit words without sign bits}
procedure plus;
begin
     if low=31744 then
        begin
             low:=$8000;
             lower:=low;
             flag:=1;
        end
        else
        begin
             low:=low+1024;
             lower:=low;
```

```
          end;
end;

{Increments the low order and trips high when above 32768}
procedure minus;
begin
     if low=$8000 then
     begin
        low:=0;
     end;
        plus;
        lower:=lower or $8000;
        if low = $8000 then
        begin
             lower:=-1024;
             low:=-1024;
             high:=high+1;
             flag:=0;
        end;
end;
Procedure filsiz;
begin
   ende:=filesize(samples);  {determine file size in
                              128 byte blocks}
   ende:=ende shr 4;        {adjust file size to 2048
                             byte blocks}
   dec:=ende;               {set high and low
                             counter file size indicators}
   if ende > 128 then
   begin
        if (ende-512) >= 0 then
        begin
             dec:=dec-512; histop:=8;
        end;
        if (dec-256) >= 0 then
        begin
             dec:=dec-256; histop:=histop+4;
        end;
        if (dec-128) >= 0 then
        begin
             dec:=dec-128; histop:=histop+2;
        end;
   end;
   if (dec-64) >= 0 then
   begin
        dec:=dec-64; histop:=histop+1;
   end;
   if (dec-32) >= 0 then
   begin
        dec:=dec-32; lostop:=$8000;
   end;
   if (dec-16) >= 0 then
```

```
   begin
        dec:=dec-16; lostop:=(lostop) or ($4000);
   end;
   if (dec-8) >= 0 ther
   begin
        dec:=dec-8; lostop:=(lostop) or ($2000);
   end;
   if (dec-4) >= 0 then
   begin
        dec:=dec-4; lostop:=(lostop) or ($1000);
   end;
   if (dec-2) >= 0 then
   begin
        dec:=dec-2; lostop:=(lostop) or ($0800);
   end;
   if (dec-1) = 0 then
      lostop:=(lostop) or ($0400);
end;          {end of procedure filsiz}
procedure copy;  {This procedure copies file from d: to C:}
begin
     assign(samples,sampname);
     reset(samples); rewrite(tram);
     filsiz;
     if histop > 2 then
     begin
         for I:=1 to ende do
         begin
              blockread(samples,buffer,16);
              blockwrite(tram,buffer,16);
         end;
     end;
     close(samples);
end;  {end of proc. copy}

procedure dispnum; {proc to dipslay the number withing
                    the block onto the screen}
       begin
       ll:=ll+1;
       window(1,24,40,25);
       textcolor(white);
       if ll=1 then
       writeln; writeln('           completed #');
       writeln; writeln('                ',ll);
       window(1,1,40,25);
       end;
procedure output;
begin
{This procedure loads from dos,filters data and
outputs data While data is output, current file
is erased and the next file is loaded from laser
disk to C: ram  }
 {open an 'untyped' file for a block read of samples}
```

```
High :=0;
Low :=0;
lower:=0;
flag :=0;
reset(tram);
assign(samples,sampname);  reset(samples);
if L=5 then L:=12;
if L=11 then L:=13;    {use only 10 of 12 filters}
L:=L-1;   {resmon addresses user functions
           from zero not one}
ftype:=($B080)or(L);
delay(1000);
for I := 1 to ende do
 begin
if histop > 2 then
 Blockread(Tram,Buffer,16)   {read first 2048 bytes}
 else
 Blockread(Samples,Buffer,16); {words & sentences
                                 no transfer}
  if L<>12 then
   begin
   if I <> 1 then
   begin
      delay(1);
      if Portw[$33C] <> $FFFF then   {filtering finished?}
           writeln('no go');
   end;
   end;
   Dcall.fun :=27;                  {initiate cmd code 11}
   Dcall.P2 :=$AC00;                {send cmd code 11}
   Dsp16;
   if Dcall.RetCode <> 0 then       {check for error}
  Derror(Dcall);
  Dcall.fun :=27;                  {send word #2 to dsp-16}
  Dcall.P2 :=High;                 {start addres, high part}
  Dsp16;
  if Dcall.RetCode <> 0 then
     Derror(Dcall);
  Dcall.fun :=27;                  {send word #3 to dsp-16}
  Dcall.P2 :=Lower;                {start address, low part}
  Dsp16;
  if Dcall.RetCode <> 0 then
     Derror(Dcall);
  Dcall.fun :=27;                  {send word #4 to dsp-16}
  Dcall.P2 :=$0000;                {high word of block length}
  Dsp16;
  if Dcall.RetCode <> 0 then
     Derror(Dcall);
  Dcall.Fun :=27;                  {send word #5 to dsp-16}
  Dcall.P2 :=blkln;                 {low word of block length}
  Dsp16;
 if Dcall.RetCode <> 0 then
```

```
    Derror(Dcall);
 {Begin transfer of actuall samples to dsp-16 data buffer}
 Dcall.fun :=28;                {send a stream of words}
 Dcall.P2 :=blkln;               {1024 words to be sent}
 Dcall.P6 :=addr(Buffer[1]);    {point to adress of samples}
 Dsp16;
 if Dcall.RetCode <> 0 then
    Derror(Dcall);
 templ:=lower;
 temph:=high;         {save address to send filter software}
 if flag = 1 then
    minus;
 if flag = 0 then
    plus;
 {must read this from dsp-16 before further communications}
 if Portw[$33C] <> $FFFF then
    writeln('Port 33CH Flagged!!!!!!!!!!!');
       if L<>12 then
begin
 {Begin user function zero in cmd code 12 to filter data}
  Dcall.Fun :=27;
  Dcall.P2 :=ftype;              {initialize cmd code #12}
   Dsp16;
   if Dcall.RetCode <> 0 then
      Derror(Dcall);
  Dcall.Fun :=27;
  Dcall.P2 :=templ;         {send low word of linear address}
  Dsp16;
  if Dcall.RetCode <> 0 then
     Derror(Dcall);
  Dcall.Fun :=27;
  Dcall.P2 :=temph;        {send high word of linear address}
  Dsp16;
  if Dcall.RetCode <> 0 then
     Derror(Dcall);
  end; {end of L<>13 if statement}
  if eof(samples) then ende:=600;
     end;
   Close(Samples);
   {Start outputting the data just received to channel B}
   if sync=1 then begin  {sync sentences and CD}
      sound(250); delay(600); nosound; sync:=0; readln;
   end;
   Dcall.Fun :=27;          {Send a word to the TMS320}
   Dcall.P2 :=$BC06;        {initiate command code 15}
   Dsp16;         {BC02 to continue loop, BC06 single shot}
   if Dcall.RetCode <> 0 then
      Derror(Dcall);
   Dcall.Fun :=27;
   Dcall.P2 :=$0000;            {High part of start address}
   Dsp16;
   if Dcall.RetCode <> 0 then
```

```
      Derror(Dcall);
   Dcall.Fun :=27;
   Dcall.P2 :=$0000;             {Low part of start address}
   Dsp16;
   if Dcall.RetCode <> 0 then
      Derror(Dcall);
   Dcall.Fun :=27;
   Dcall.P2 :=histop;            {High part of end address}
   Dsp16;
   if Dcall.RetCode <> 0 then
      Derror(Dcall);
   Dcall.Fun :=27;
   Dcall.P2 :=lostop;            {Low part of end address}
   Dsp16;
   if Dcall.RetCode <>0 then
      Derror(Dcall);
   if Portw[$33C] <> $FFFF then
      writeln('port 33CH flagged!!!!!!');
   if sampname2 <> 'd:end  ' then
   begin
      sampname:=sampname2;
      L:=L2; SNR:=SNR2;
      if (histop > 2) and (sampname <> 'd:bend ') then
      begin
      copy; writeln('copy!!!!!!!!!!!!!!!!!!');
      end;        {copy only for CD}
   end;
   if sampname2 ='d:end  ' then sampname:='d:end    ';
   if sampname = 'd:end  ' then delay(23000);
   {if CDs are over then avoid}
   if histop > 2 then delay(6000) else delay(4500);
   {handshake error}
end;  {end of procedure output}
procedure woutput;
begin
{This procedure loads carrier phrase from laser disk, }
{half a second of silence and the specified word. Only}
{the word is filtered. The buffer is then output.     }
   {open an 'untyped' file for a block read of samples}
   reset(carry);
   assign(samples,sampname); reset(samples);
   High :=0;
   Low :=0;
   lower:=0;
   flag :=0;
   ende:=72;
   if L=5 then L:=12;
   if L=11 then L:=13;   {use only 10 of 12 filters}
   L:=L-1;    {resmon addresses user functions from
                zero not one}
   ftype:=($B080)or(L);
   delay(100);
```

```
for I := 1 to ende do
    begin
    if (I > 36) and (loadcp = 1) then
         Blockread(carry,buffer,16);
         {get carrier phrase}
    if I <= 36 then
            Blockread(Samples,Buffer,16);
            {go get the words}
    if L<>12 then
    begin
    if I <> 1 then
    begin
         delay(1);
         if Portw[$33C] <> $FFFF then
         {filtering finished?}
             writeln('no go');
    end;
    end;
    {skip loading of carrier phrase if not the
    beginning of 14 word block}
    if ((I > 36) and (loadcp = 1)) or (I <= 36) then
    begin
    Dcall.fun :=27;                  {initiate c? 1 code 11}
    Dcall.P2 :=$AC00;                {send cmd code 11}
    Dsp16;
    if Dcall.RetCode <> 0 then       {check for error}
       Derror(Dcall);
    Dcall.fun :=27;             {send word #2 to dsp-16}
    Dcall.P2 :=High;            {start addres, high part}
    Dsp16;
    if Dcall.RetCode <> 0 then
       Derror(Dcall);
    Dcall.fun :=27;             {send word #3 to dsp-16}
    Dcall.P2 :=Lower;           {start address, low part}
    Dsp16;
    if Dcall.RetCode <> 0 then
       Derror(Dcall);
    Dcall.fun :=27;             {send word #4 to dsp-16}
    Dcall.P2 :=$0000;           {high word of block length}
    Dsp16;
    if Dcall.RetCode <> 0 then
       Derror(Dcall);
    Dcall.Fun :=27;             {send word #5 to dsp-16}
    Dcall.P2 :=blkln;            {low word of block length}
    Dsp16;
    if Dcall.RetCode <> 0 then
       Derror(Dcall);
{Begin transfer of actuall samples to dsp-16 data buffer}
     Dcall.fun :=28;             {send a stream of words}
     Dcall.P2 :=blkln;           {1024 words to be sent}
   Dcall.P6 :=addr(Buffer[1]);{point to adress of samples}
     Dsp16;
```

```
        if Dcall.RetCode <> 0 then
           Derror(Dcall);
        templ:=lower;
        temph:=high;
        {save address to send filter software}
        if flag = 1 then
           minus;
        if flag = 0 then
           plus;
 {must read this from dsp-16 before further communications}
        if Portw[$33C] <> $FFFF then
           writeln('Port 33CH Flagged!!!!!!!!!!!');
        if L<>12 then
        begin
        if I <= 36 then
        begin
{Begin user function zero in cmd code 12 to filter data}
          Dcall.Fun :=27;
          Dcall.P2 :=ftype;        {initialize cmd code #12}
          Dsp16;
          if Dcall.RetCode <> 0 then
             Derror(Dcall);
          Dcall.Fun :=27;
          Dcall.P2 :=templ;  {send low word of linear
address}
          Dsp16;
          if Dcall.RetCode <> 0 then
             Derror(Dcall);
          Dcall.Fun :=27;
          Dcall.P2 :=temph;{send high word of linear address}
          Dsp16;
          if Dcall.RetCode <> 0 then
             Derror(Dcall);
          end; {end of I > 40 statement}
          end; {end of L<>13 if statement}
    end; {end of skip load loop}
    if (I>36) and (loadcp=0) then
    begin
         if flag = 1 then minus;
         if flag = 0 then plus;
    end;
    end; {end of I do loop}
    loadcp:=0;
    Close(Samples);
{The outputting of the data needs to be done in reverse,
i.e. the word loaded in the buffer before the carrier
phrase since the tms32020 filter routine initializes
itself when it recieves a buffer address of  zero. BEGIN
output routine!!!}
    if sync=1 then begin
    sound(250); delay(600); nosound; readln; sync:=0;
    end;
```

```
Dcall.Fun :=27;          {Send a word to the TMS320}
Dcall.P2 :=$BC06;         {initiate command code 15}
Dsp16;          {BC02 to continue loop, BC06 single shot}
if Dcall.RetCode <> 0 then
   Derror(Dcall);
Dcall.Fun :=27;
Dcall.P2 :=$0000;        {High part of start address}
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
Dcall.Fun :=27;
Dcall.P2 :=$9000;         {Low part of start address}
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
Dcall.Fun :=27;
Dcall.P2 :=1;         {High part of end address}
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
Dcall.Fun :=27;
Dcall.P2 :=$2000;             {Low part of end address}
Dsp16;
if Dcall.RetCode <>0 then
   Derror(Dcall);
if Portw[$33C] <> $FFFF then
   writeln('port 33CH flagged!!!!!!');
delay(2200);
{Start outputting the word section of output}
Dcall.Fun:=18;
Dcall.P2:=1;
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
Dcall.Fun :=27;              {Send a word to the TMS320}
Dcall.P2 :=$BC06;         {initiate command code 15}
Dsp16;         BC02 to continue loop, BC06 single shot}
if Dcall.RetCode <> 0 then
   Derror(Dcall);
Dcall.Fun :=27;
Dcall.P2 :=$0000;            {High part of start address}
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
Dcall.Fun :=27;
Dcall.P2 :=$0000;             {Low part of start address}
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
Dcall.Fun :=27;
Dcall.P2 :=0;             {High part of end address}
Dsp16;
```

```
   if Dcall.RetCode <> 0 then
      Derror(Dcall);
   Dcall.Fun :=27;
   Dcall.P2 :=$9000;            {Low part of end address}
   Dsp16;
   if Dcall.RetCode <>0 then
      Derror(Dcall);
   if Portw[$33C] <> $FFFF then
      writeln('port 33CH flagged!!!!!!');
   if sampname2 <> 'd:end' then
   begin
      sampname:=sampname2;
      L:=L2; SNR:=SNR2;
   end;
   if sampname2 ='d:end' then sampname:='d:end';
   delay(2200);
   close(carry);
   dispnum;
end;  {end of procedure woutput}
procedure Bblock;
begin
    clrscr;
    sync:=1;
    ll:=0;
    textmode(c40);
    for I:=1 to 10 do
    begin
         writeln;
    end;
    writeln('           SNR SETTING IS ',SNR);
    writeln; writeln;
    writeln('          enter to start block');
    writeln; writeln;
    writeln('     THIS BLOCK BEGINS WITH ',sampname);
    writeln; writeln; writeln;
    loadcp:=1;
    if histop > 2 then copy;
end;    {end procedure bblock}

begin
   ErrFlg := False;
   ClrScr;
   GotoXY(30,3);
   Write('DSP-16 Interface');
   GotoXY(36,5);
   Write('for');
   GotoXY(30,7);
   Write('Turbo Pascal 3.0');
   GotoXY(1,9);
   { Check for an interrupt handler installed }
   writeln('Checking for resident DSP-16 driver.');
   if (IntOfs = 0) or (IntSeg = 0) then
```

```
begin
   Sound(250);
   Delay(200);
   NoSound;
   Writeln('No DSP-16 handler found.
   At DOS prompt, type: DSP11TPA.');
   Exit;
end;
{Call an invalid func to check for DSP-16 handler. }
Dcall.Fun := 0;
Dsp16;
if Dcall.RetCode <> 101 then
begin
   Sound(250);
   Delay(200);
   NoSound;
   Writeln('Improper interface loaded.
   Check for conflicting equipment.');
   Exit;
end;
writeln('Enter init to initialize');
readln(init);
if init = 'init' then
begin
Writeln('Setting address of DSP-16.');
Dcall.Fun := 2;              { Function Number }
Dcall.P2 := DspAddr;         { I/O Base Address }
Dcall.P3 := DspPage;         { Memory Page }
Dcall.P4 := DspInt;          { Interrupt Enable/Number }
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
Writeln('Initializing DSP-16 hardware.');
Dcall.Fun := 1;          { Function Number }
Dcall.P2 := 256;         { Size of buffer in kbytes }
Dcall.P3 := 16;          { Size of program RAM in kbytes }
Dcall.P10^ := DefFile;   { Program to Upload }
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
{set up buffer as 1M from the default of 256K}
Dcall.fun :=27;
Dcall.P2 :=$9CF1;
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
end;
Writeln('DSP-16 Driver version ',
Dcall.P3:1,'.',Dcall.P4:1,'.');
Writeln('Setting sample rate to',
1000.0/(DefSr*0.8):7:3,' kHz');
Dcall.Fun := 4;      { Function Number }
```

```
Dcall.P2 := DefSr;   { Input Sample Rate Divider }
Dcall.P3 := DefSr;   { Output Sample Rate Divider }
Dcall.P4 := 0;       { Use Default Sample/Hold Delay }
Dcall.P5 := 0;       { No I/O synchronization }
Dsp16;
if Dcall.RetCode <> 0 then
   Derror(Dcall);
{begin actual main part of main program}
sync:=0;
assign(tram,tempram);
writeln('enter the subject file to be used');
readln(ordername);
assign(order,ordername); reset(order);
assign(carry,cphrase);
readln(order,sampname,L,SNR);
sampname:=concat('d:',sampname);
I:=0;          {begin routine allow starting in middle}
writeln('enter a 1 to start from other
than the beggining');
readln(I);
if I=1 then
begin
writeln('enter the block to begin with');
readln(I);
{J:=pos('w',sampname);
if J<>0 then num:=14;
J:=pos('s',sampname);
if J<>0 then num:=4;
J:=pos('c',sampname);
if J<>0 then num:=1;}
for J:=1 to I-1 do
begin
 while sampname <> 'd:bend ' do
 begin
  writeln(I,' ',sampname);
  readln(order,sampname,L,SNR);
  sampname:=concat('d:',sampname);
  end;
  readln(order,sampname,L,SNR);
  sampname:=concat('d:',sampname);
  end;
end;
loadcp:=0;
bblock;
copy;     {also calls filsiz}
readln(order,sampname2,L2,SNR2);
sampname2:=concat('d:',sampname2);
while sampname <> 'd:end  ' do
begin
  if sampname = 'd:bend ' then
  begin
       sampname:=sampname2; L:=L2; SNR:=SNR2;
```

```
        readln(order,sampname2,L2,SNR2);
        sampname2:=concat('d:',sampname2);
        bblock;
   end;
   lostop:=0; histop:=0;   {initialize file size counters}
   assign(samples,sampname); reset(samples);
   filsis;
   close(samples);
   I:=pos('w',sampname);
   if  I <> 0 then
     woutput
     else
     output;
   readln(order,sampname2,L2,SNR2);
   sampname2:=concat('d:',sampname2);
   pause:=keypressed;
   if keypressed = true then
   begin
    writeln('A key was pressed, program currently paused');
    readln;
    pause:=false;
   end;
 end;  {end of while loop};
 close(order); close(tran);  close(carry);
 textmode;
end.
```

TMS32020 assembler program to filter speech with the Ariel DSP-16

```
;This program is a general filter routine. all filters
;have attenuation of 60 db. Max. passband ripple is .1
;db, sampling rate  is 18.1159 kHz  The program is
;currently defined as user function one thru twelve.
;Two parameters are passed to it by the high level
;program, the high word of the start address and the low
;word of the start ad

     YN:          EQU     96
     Counter      EQU     97
     Bufdat:      EQU     98
     Dummy:       EQU     99
     Stat:        EQU     100
     FLEN:        EQU     101
     RAMTYPE:     EQU     102
     BOTB1:       EQU     103
     ODD:         EQU     104
                  ORG     1024
     START_1:
                  LDPK    0
                  ZAC
```

```
        SACL   COUNTER  ;zero counter
        LACK   6        ;load acclo w/ t shift
        SACL   RAMTYPF  ;set up t reg shift
        LACK   1        ;odd value for refresh
        SACL   ODD
        RET
ONE:    CALL   START_1  ;go set counter for lp400
        LAC    RM_USER1 ;load lo word addr in acclo
        BGZ    FILTER   ;initialize?
        LAC    RM_USER2 ;ld hiword addr in acchi
        BGZ    FILTER   ;initialize?
        CNFD
        LACK   165      ;load filter lenght
        SACL   FLEN
        LALK   859       ;load bottom of B1
        SACL   BOTB1
        LARP   0
        LRLK   AR0,200H  ;point to block B0
        RPT    FLEN
        BLKP   3000,*+   ;load coefs into B0
        CNFP
        ZAC
        LAR    AR0,BOTB1  ;bottom of B1
        RPT    FLEN
        SACL   *+        ;load B1 with zeroes
        B      FILTER    ;go execute filter
TWO:    CALL   START_1   ;go set counter for lp650
        LAC    RM_USER1 ;load lo word addr in acclo
        BGZ    FILTER   ;initialize?
        LAC    RM_USER2 ;load hi word addr in acchi
        BGZ    FILTER   ;initialize?
        CNFD
        LACK   103      ;load filter lenght
        SACL   FLEN
        LALK   921      ;load bottom of B1
        SACL   BOTB1
        LARP   0
        LRLK   AR0,200H ;point to block B0
        RPT    FLEN
        BLKP   3165,*+  ;load coefs into B0
        CNFP
        ZAC
        LAR    AR0,BOTB1 ;bottom of B1
        RPT    FLEN
        SACL   *+        ;load B1 with zeroes
        B      FILTER    ;go execute filter
THREE:  CALL   START_1   ;go set counter  for lp920
        LAC    RM_USER1  ;ld lo wd addr in acclo
        BGZ    FILTER    ;initialize?
        LAC    RM_USER2  ;ld hi word addr in acchi
        BGZ    FILTER    ;initialize?
        CNFD
```

```
        LACK    73          ;load filter lenght
        SACL    FLEN
        LALK    951         ;load bottom of B1
        SACL    BOTB1
        LARP    0
        LRLK    AR0,200H    ;point to block B0
        RPT     FLEN
        BLKP    3269,*+     ;load coefs into B0
        CNFP
        ZAC
        LAR     AR0,BOTB1   ;bottom of B1
        RPT     FLEN
        SACL    *+          ;load B1 with zeroes
        B       FILTER      ;go execute filter
FOUR:   CALL    START_1     ;go set cnter for 1p1400
        LAC     RM_USER1    ;ld lo word addr in acclo
        BGZ     FILTER      ;initialize?
        LAC     RM_USER2    ;ld hi word addr in acchi
        BGZ     FILTER      ;initialize?
        CNFD
        LACK    47          ;load filter lenght
        SACL    FLEN
        LALK    977         ;load bottom of B1
        SACL    BOTB1
        LARP    0
        LRLK    AR0,200H    ;point to block B0
        RPT     FLEN
        BLKP    3343,*+     ;load coefs into B0
```

```
            CNFP
            ZAC
            LAR     AR0,BOTB1 ;bottom of B1
            RPT     FLEN
            SACL    *+        ;load B1 with zeroes
            B       FILTER    ;go execute filter
    FIVE:   CALL    START_1   ;go set counter for lp3000
            LAC     RM_USER1  ;load lo word addr in
acclo
            BGZ     FILTER    ;initialize?
            LAC     RM_USER2  ;load hi word addr in
acchi
            BGZ     FILTER    ;initialize?
            CNFD
            LACK    23        ;load filter lenght
            SACL    FLEN
            LALK    1001      ;load bottom of B1
            SACL    BOTB1
            LARP    0
            LRLK    AR0,200H  ;point to block B0
            RPT     FLEN
            BLKP    3390,*+   ;load coefs into B0
            CNFP
            ZAC
            LAR     AR0,BOTB1 ;bottom of B1
            RPT     FLEN
            B       FILTER    ;go execute filter
    SIX:    CALL    START_1   ;go set cnter for lp4500
            LAC     RM_USER1  ;ld lo wd addr in acclo
            BGZ     FILTER    ;initialize?
            LAC     RM_USER2  ;ld hi wd addr in acchi
            BGZ     FILTER    ;initialize?
            CNFD
            LACK    15        ;load filter lenght
            SACL    FLEN
            LALK    1009      ;load bottom of B1
            SACL    BOTB1
            LARP    0
            LRLK    AR0,200H  ;point to block B0
            RPT     FLEN
            BLKP    3414,*+    ;load coefs into B0
            CNFP
            ZAC
            LAR     AR0,BOTB1  ;bottom of B1
            RPT     FLEN
            SACL    *+         ;load B1 with zeroes
            B       FILTER     ;go execute filter
    SEVEN:  CALL    START_1    ;go set counter for hp
920
            LAC     RM_USER1   ;ld lo wd addr in acclo
            BGZ     FILTER     ;initialize?
            LAC     RM_USER2   ;ld hi wd addr in acchi
```

```
        BGZ     FILTER      ;initialize?
        CNFD
        LACK    141         ;load filter lenght
        SACL    FLEN
        LALK    883         ;load bottom of B1
        SACL    BOTB1
        LARP    0
        LRLK    AR0,200H    ;point to block B0
        RPT     FLEN
        BLKP    3632,*+     ;load coefs into B0
        CNFP
        ZAC
        LAR     AR0,BOTB1   ;bottom of B1
        RPT     FLEN
        SACL    *+          ;load B1 with zeroes
        B       FILTER      ;go execute filter
EIGHT:  CALL    START_1     ;go set cnter for hp 1400
        LAC     RM_USER1    ;ld lo wd addr in acclo
        BGZ     FILTER      ;initialize?
        LAC     RM_USER2    ;ld hi wd addr in acchi
        BGZ     FILTER      ;initialize?
        CNFD
        LACK    93         ;load filter lenght
        SACL    FLEN
        LALK    931        ;load bottom of B1
        SACL    BOTB1
        LARP    0
        LRLK    AR0,200H ;point to block B0
        RPT     FLEN
        BLKP    3774,*+  ;load coefs into B0
        CNFP
        ZAC
        LAR     AR0,BOTB1 ;bottom of B1
        RPT     FLEN
        SACL    *+          ;load B1 with zeroes
        B       FILTER      ;go execute filter
NINE:   CALL    START_1     ;set cnter for hp 2200
        LAC     RM_USER1    ;ld lo wd addr in acclo
        BGZ     FILTER      ;initialize?
        LAC     RM_USER2    ;ld hi wd addr in acchi
        BGZ     FILTER      ;initialize?
        CNFD
        LACK    59          ;load filter lenght
        SACL    FLEN
        LALK    965         ;load bottom of B1
        SACL    BOTB1
        LARP    0
        LRLK    AR0,200H    ;point to block B0
        RPT     FLEN
        BLKP    3866,*+     ;load coefs into B0
        CNFP
        ZAC
```

```
                LAR     AR0,BOTB1 ;bottom of B1
                RPT     FLEN
                SACL    *+        ;load B1 with zeroes
                B       FILTER    ;go execute filter
      TEN.      CALL    START_1   ;set cnter for hp 3500
                LAC     RM_USER1  ;load lo word addr in
acclo
                BGZ     FILTER    ;initialize?
                LAC     RM_USER2  ;ld hi wd addr in acchi
                BGZ     FILTER    ;initialize?
                CNFD
                LACK    43        ;load filter lenght
                SACL    FLEN
                LALK    981       ;load bottom of B1
                SACL    BOTB1
                LARP    0
                LRLK    AR0,200H  ;point to block B0
                RPT     FLEN
                BLKP    3926,*+   ;load coefs into B0
                CNFP
                ZAC
                LAR     AR0,BOTB1 ;bottom of B1
                RPT     FLEN
                SACL    *+        ;load B1 with zeroes
                B       FILTER    ;go execute filter
      ELEVEN:   CALL    START_1   ;set cnter for hp 4500
                LAC     RM_USER1  ;ld lo wd addr in acclo
                BGZ     FILTER    ;initialize?
                LAC     RM_USER2  ;ld hi wd addr in acchi
                BGZ     FILTER    ;initialize?
                CNFD
                LACK    29        ;load filter lenght
                SACL    FLEN
                LALK    995       ;load bottom of B1
                SACL    BOTB1
                LARP    0
                LRLK    AR0,200H ;point to block B0
                RPT     FLEN
                BLKP    3970,*+  ;load coefs into B0
                CNFP
                ZAC
                LAR     AR0,BOTB1 ;bottom of B1
                RPT     FLEN
                SACL    *+        ;load B1 with zeroes
                B       FILTER    ;go execute filter
      TWELVE:   CALL    START_1   ;set cnter for hp 5500
                LAC     RM_USER1  ;ld lo wd addr in acclo
                BGZ     FILTER    ;initialize?
                LAC     RM_USER2 ;load hi word addr in acchi
                BGZ     FILTER    ;initialize?
                CNFD
                LACK    23        ;load filter lenght
```

```
                SACL    FLEN
                LALK    1001      ;load bottom of B1
                SACL    BOTB1
                LARP    0
                LRLK    AR0,200H  ;point to block B0
                RPT     FLEN
                BLKP    4000,*+   ;load coefs into B0
                CNFP
                ZAC
                LAR     AR0,BOTB1 ;bottom of B1
                RPT     FLEN
                SACL    *+        ;load B1 with zeroes
                B       FILTER    ;go execute filter
FILTER:         RSXM              ;sign extension mode off
                OUT     ODD,11    ;turn refresh off
                IN      DUMMY,8   ;dummy rd,shut of refresh
                LT      RAMTYPE   ;load T with shift factor
                LACT    RM_USER1
                ;bits 10.15 shifted to bits 0.5
                                  ;of acchi
                SACH    RM_USER3  ;save
                ZALS    RM_USER3  ;reload in acclo
                ADDT    RM_USER2
                 ;merge in bits 0..3 of rm_user2 as
                SACL    RM_USER3  ;bits 6..9 of cas address
                OUT     RM_USER1, ;send ras address
                OUT     RM_USER3,9 ;send cas address
                IN      BUFDAT,8  ;get data
                OUT     RAMTYPE,11;turn refresh back on
                LAR     AR1,BOTB1 ;point to bottom of B1
                LARP    AR1
                LAC     Bufdat    ;load x(n) into B1
                SACL    *,0       ;via accumulator
                LRLK    AR1,3FFH
                MPYK    0         ;set P reg. to zero
                ZAC               ;clear accumulator
                RPT     FLEN      ;execute the filter
                MACD    0FF00H,*-
                APAC
                SACH    YN,1
                RSXM
                OUT     ODD,11    ;refresh off
                IN      DUMMY,8   ;addr already formatted in
                OUT     RM_USER1,9  ;rm_user one and three
                OUT     RM_USER3,9
                OUT     YN,8      ;put flterd data n buffer
                NOP
                OUT     RAMTYPE,11 ;turn refresh back on
                LAC     RM_USER1
                ADLK    1          ;increment to next
address
                SACL    RM_USER1
```

```
        LAC     Counter   ;increment counter
        ADLK    1
        SACL    Counter   ;store incremented counter
        LALK    1024      ;can 1024 points processed
        SUBS    Counter
        BNZ     FILTER    ;if not dc again
        LALK    0FFFFH    ;send FFFF to port 33ch
        SACL    Dummy
WAIT:   IN      Stat,14   ;test status port
        BIT     Stat,1    ;can something be sent
        BBNZ    WAIT      ;continue if not
        OUT     Dummy,15  ;else send the data
        RET
        ORG       3000  ;lp400
        DW    0FFFDH,0FFFDH,0FFFDH,0FFFDH,0FFFDH
        DW    0FFFFH,0FFFFH,0001H,0003H,0005H
        DW    0007H,000AH,000DH,000FH,0011H
        DW    0013H,0013H,0013H,0011H,000EH
        DW    0009H,0004H,0FFFDH,0FFF5H,0FFEDH
        DW    0FFE3H,0FFDBH,0FFD1H,0FFCBH,0FFC5H
        DW    0FFC1H,0FFC1H,0FFC3H,0FFCAH,0FFD3H
        DW    0FFE1H,0FFF3H,0006H,001CH,0034H
        DW    004DH,0064H,0079H,008CH,0099H
        DW    00A1H,00A1H,009AH,008AH,0071H
        DW    004FH,0027H,0FFF7H,0FFC1H,0FF87H
        DW    0FF4DH,0FF14H,0FEDFH,0FEB3H,0FE90H
        DW    0FE7BH,0FE76H,0FE83H,0FEA5H,0FEDDH
        DW    0FF2DH,0FF92H,000DH,009CH,013DH
        DW    01EEH,02A9H,036CH,0431H,04F3H
        DW    05ADH,065AH,06F6H,077DH,07E9H
        DW    0839H,086AH,087BH,086AH,0839H
        DW    07E9H,077DH,06F6H,065AH,05ADH
        DW    04F3H,0431H,036CH,02A9H,01EEH
        DW    013DH,009CH,000DH,0FF92H,0FF2DH
        DW    0FEDDH,0FEA5H,0FE83H,0FE76H,0FE7BH
        DW    0FE90H,0FEB3H,0FEDFH,0FF14H,0FF4DH
        DW    0FF87H,0FFC1H,0FFF7H,0027H,004FH
        DW    0071H,008AH,009AH,00A1H,00A1H
        DW    0099H,008CH,0079H,0064H,004DH
        DW    0034H,001CH,0006H,0FFF3H,0FFE1H
        DW    0FFD3H,0FFCAH,0FFC3H,0FFC1H,0FFC1H
        DW    0FFC5H,0FFCBH,0FFD1H,0FFDBH,0FFE3H
        DW    0FFEDH,0FFF5H,0FFFDH,0004H,0009H
        DW    000EH,0011H,0013H,0013H,0013H
        DW    0011H,000FH,000DH,000AH,0007H
        DW    0005H,0003H,0001H,0FFFFH,0FFFFH
        DW    0FFFDH,0FFFDH,0FFFDH,0FFFDH,0FFFDH
        ORG       3165  ;lp650
        DW    0FFFCH,0FFFBH,0FFFBH,0FFFBH,0FFFEH
        DW    0002H,0008H,000FH,0017H,001DH
        DW    0021H,0020H,001AH,000FH,0FFFDH
        DW    0FFE7H,0FFCEH,0FFB5H,0FFA2H,0FF97H
```

```
DW   0FF99H,0FFA9H,0FFC9H,0FFF9H,0033H
DW   0075H,00B3H,00E7H,0105H,0107H
DW   00E7H,00A0H,0035H,0FFAFH,0FF17H
DW   0FE7FH,0FDF9H,0FD9DH,0FD7DH,0FDABH
DW   0FE35H,0FF1FH,0069H,0205H,03E1H
DW   05E1H,07E5H,09C9H,0B6EH,0CB3H
DW   0D81H,0DC7H,0D81H,0CB3H,0B6EH
DW   09C9H,07E5H,05E1H,03E1H,0205H
DW   0069H,0FF1FH,0FE35H,0FDABH,0FD7DH
DW   0FD9DH,0FDF9H,0FE7FH,0FF17H,0FFAFH
DW   0035H,00A0H,00E7H,0107H,0105H
DW   00E7H,00B3H,0075H,0033H,0FFF9H
DW   0FFC9H,0FFA9H,0FF99H,0FF97H,0FFA2H
DW   0FFB5H,0FFCEH,0FFE7H,0FFFDH,000FH
DW   001AH,0020H,0021H,001DH,0017H
DW   000FH,0008H,0002H,0FFFEH,0FFFBH
DW   0FFFBH,0FFFBH,0FFFCH
ORG    3269 ;lp920
DW   0FFFBH,0FFF9H,0FFF9H,0FFFFH,0009H
DW   0017H,0025H,002EH,002BH,0019H
DW   0FFF6H,0FFC7H,0FF96H,0FF73H,0FF6DH
DW   0FF8FH,0FFDFH,0053H,00D4H,0142H
DW   0178H,0155H,00CAH,0FFDDH,0FEB3H
DW   0FD88H,0FCADH,0FC77H,0FD29H,0FEE8H
DW   01AFH,0547H,094AH,0D38H,1086H
DW   12BAH,1380H,12BAH,1086H,0D38H
DW   094AH,0547H,01AFH,0FEE8H,0FD29H
DW   0FC77H,0FCADH,0FD88H,0FEB3H,0FFDDH
DW   00CAH,0155H,0178H,0142H,00D4H
DW   0053H,0FFDFH,0FF8FH,0FF6DH,0FF73H
DW   0FF96H,0FFC7H,0FFF6H,0019H,002BH
DW   002EH.0025H,0017H,0009H,0FFFFH
DW   0FFF9H,0FFF9H,0FFFBH
ORG     3343                ;lp1400
DW   0FFF9H,0FFFBH,000BH,0027H,003DH
DW   002DH,0FFEBH,0FF83H,0FF33H,0FF49H
DW   0FFF2H,0105H,01F7H,0211H,00CFH
DW   0FE5FH,0FBC5H,0FAA5H,0FC8BH,0221H
DW   0A9FH,13D9H,1AFDH,1DADH,1AFDH
DW   13D9H,0A9FH,0221H,0FC8BH,0FAA5H
DW   0FBC5H,0FE5FH,00CFH,0211H,01F7H
DW   0105H,0FFF2H,0FF49H,0FF33H,0FF83H
DW   0FFEBH,002DH,003DH,0027H,000BH
DW   0FFFBH,0FFF9H
ORG     3390  ;lp3000
DW   0FFEDH,0006H,008DH,0FFEBH,0FE2BH
DW   002EH,04AAH,0FFB7H,0F4D1H,0060H
DW   27E3H,3F97H,27E3H,0060H,0F4D1H
DW   0FFB7H,04AAH,002EH,0FE2BH,0FFEBH
DW   008DH,0006H,0FFEDH
ORG     3414   ;lp4500
DW   0FFEDH,00A0H,0FEA1H,003FH,059AH
```

```
DW   0EF8CH,1BBFH,5F63H,1BBFH,0EF8CH
DW   059AH,003FH,0FEA1H,00A0H,0FFEDH
ORG     3632                    ;hp920
DW   0003H,0003H,0003H,0001H,0001H
DW   0FFFFH,0FFFDH,0FFFAH,0FFF7H,0FFF3H
DW   0FFF1H,0FFEEH,0FFEDH,0FFECH,0FFEDH
DW   0FFF1H,0FFF5H,0FFFCH,0005H,000FH
DW   001AH,0025H,0031H,003AH,0041H
DW   0045H,0045H,003FH,0035H,0025H
DW   000FH,0FFF6H,0FFD9H,0FFB9H,0FF9BH
DW   0FF7DH,0FF65H,0FF53H,0FF4AH,0FF4CH
DW   0FF5AH,0FF75H,0FF9DH,0FFD2H,0011H
DW   0058H,00A3H,00EFH,0137H,0175H
DW   01A4H,01BFH,01C0H,01A5H,0169H
DW   010BH,008BH,0FFEAH,0FF29H,0FE4FH
DW   0FD5FH,0FC61H,0FB5DH,0FA5CH,0F966H
DW   0F884H,0F7BEH,0F71CH,0F6A3H,0F659H
DW   763FH,0F659H,0F6A3H,0F71CH,0F7BEH
DW   0F884H,0F966H,0FA5CH,0FB5DH,0FC61H
DW   0FD5FH,0FE4FH,0FF29H,0FFEAH,008BH
DW   010BH,0169H,01A5H,01C0H,01BFH
DW   01A4H,0175H,0137H,00EFH,00A3H
DW   0058H,0011H,0FFD2H,0FF9DH,0FF75H
DW   0FF5AH,0FF4CH,0FF4AH,0FF53H,0FF65H
DW   0FF7DH,0FF9BH,0FFB9H,0FFD9H,0FFF6H
DW   000FH,0025H,0035H,003FH,0045H
DW   0045H,0041H,003AH,0031H,0025H
DW   001AH,000FH,0005H,0FFFCH,0FFF5H
DW   0FFF1H,0FFEDH,0FFECH,0FFEDH,0FFEEH
DW   0FFF1H,0FFF3H,0FFF7H,0FFFAH,0FFFDH
DW   0FFFFH,0001H,0001H,0003H,0003H
DW   0003H
ORG      3774  ;hp1400
DW   0003H,0004H,0003H,0FFFFH,0FFFBH
DW   0FFF3H,0FFEDH,0FFE5H,0FFE2H,0FFE3H
DW   0FFE9H,0FFF7H,000BH,0024H,003FH
DW   0056H,0066H,0069H,005BH,003AH
DW   0007H,0FFC6H,0FF7EH,0FF39H,0FF05H
DW   0FEEBH,0FEF8H,0FF31H,0FF99H
DW   0027H,00D1H,0181H,021DH,028BH
DW   02ADH,026DH,01BBH,008EH,0FEEFH
DW   0FCF1H,0FAB1H,0F857H,0F613H,0F413H
DW   0F281H,0F181H,7129H,0F181H,0F281H
DW   0F413H,0F613H,0F857H,0FAB1H,0FCF1H
DW   0FEEFH,008EH,01BBH,026DH,02ADH
DW   028BH,021DH,0181H,00D1H,0027H
DW   0FF99H,0FF31H,0FEF8H,0FEEBH,0FF05H
DW   0FF39H,0FF7EH,0FFC6H,0007H,003AH
DW   005BH,0069H,0066H,0056H,003FH
DW   0024H,000BH,0FFF7H,0FFE9H,0FFE3H
DW   0FFE2H,0FFE5H,0FFEDH,0FFF3H,0FFFBH
DW   0FFFFH,0003H,0004H,0003H
```

```
ORG  3866                        ;HP 2200
DW   0005H,0003H,0FFFBH,0FFEDH,0FFDBH
DW   0FFD3H,0FFDBH,0FFFFH,0037H,0075H
DW   009FH,0095H,0046H,0FFB6H,0FF07H
DW   0FE79H,0FE55H,0FED1H,0FFF9H,0195H
DW   032FH,0423H,03CEH,01C0H,0FDEAH
DW   0F8B3H,0F2EDH,0EDADH,0EA01H,68AFH
DW   0EA01H,0EDADH,0F2EDH,0F8B3H,0FDEAH
DW   01C0H,03CEH,0423H,032FH,0195H
DW   0FFF9H,0FED1H,0FE55H,0FE79H,0FF07H
DW   0FFB6H,0046H,0095H,009FH,0075H
DW   0037H,0FFFFH,0FFDBH,0FFD3H,0FFDBH
DW   0FFEDH,0FFFBH,0003H,0005H
ORG     3926                     ;hp 3500
DW   0006H,0FFFEH,0FFE6H,0FFCBH,0FFCDH
DW   0009H,0075H,00CEH,00B3H,0FFEBH
DW   0FEB3H,0FDCDH,0FE22H,0023H,032DH
DW   057CH,04D9H,0FFD1H,0F6C1H,0EC19H
DW   0E381H,6034H,0E381H,0EC19H,0F6C1H
DW   0FFD1H,04D9H,057CH,032DH,0023H
DW   0FE22H,0FDCDH,0FEB3H,0FFEBH,00B3H
DW   00CEH,0075H,0009H,0FFCDH,0FFCBH
DW   0FFE6H,0FFFEH,0006H
ORG   3970                       ;HP 4500
DW   0009H,0FFEDH,0FFB0H,0FFD5H,00AFH
DW   013FH,0FFE1H,0FD39H,0FD33H,026DH
DW   083AH,045DH,0F221H,0DAF7H,504FH
DW   0DAF7H,0F221H,045DH,083AH,026DH
DW   0FD33H,0FD39H,0FFE1H,013FH,00AFH
DW   0FFD5H,0FFB0H,0FFEDH,0009H
ORG   4000                       ;HP 5500
DW   0001H,0FFC3H,0FFD5H,00F3H,0105H
DW   0FDC7H,0FC6EH,03D1H,0A37H,0FAD4H
DW   0D881H,45B5H,0D881H,0FAD4H,0A37H
DW   03D1H,0FC6EH,0FDC7H,0105H,00F3H
DW   0FFD5H,0FFC3H,0001H
```

Appendix E

EXAMPLE OF ARCSINE TRANSFORMATION

Example of the arcsine transformation for the experimental condition, low pass filter 920 Hz and S/N of 0 dB. The tabled data are for sentences.

Subject #	Raw Data	Xformed Data
1	67.5	1.93
2	22.5	0.99
3	45.0	1.47
4	67.5	1.93
5	30.0	1.12
6	22.5	0.99
7	15.0	0.80
8	37.5	1.32
9	7.5	0.55
10	22.5	0.99
11	22.5	0.99
12	0.0	0.00
13	37.5	1.32
14	37.5	1.32
15	60.0	1.77
16	75.0	2.10
17	30.0	1.16
18	52.5	1.62
19	30.0	1.16
20	30.0	1.16
21	7.5	0.55
22	60.0	1.77
23	15.0	0.80
24	7.5	0.55
% Mean	33.4	31.1
% S.D.	20.7	6.1

The above tabled % scores for the mean transformed conditions were inverse transformed in order to demonstrate the beneficial effects upon the standard deviation (S.D.) of the arcsine transform. The transform is

$$Y' = 2 \times \sin^{-1} \sqrt{\frac{s}{100}}$$

where Y' is the transformed score and s is the SIS in percent. The inverse arcsine transform is simply

$$s = \sin^2\left(\frac{Y'}{2}\right) \times 100$$

Appendix F

SUBJECT INSTRUCTIONS

INSTRUCTIONS

There will be two separate parts to this session. The first part will be a chance for you to practice the tasks that will be asked of you throughout the experiment. The second part will begin the actual experiment. Please read all instructions carefully!

Part I. Practice

a. Words You will be presented with two sets of 14 words. Each word will be preceeded by the phrase "will you write". In each set you will be provided with time to identify each individual word. Guessing is encouraged. If you cannot recognize a words, please draw a line thru that item. At the end of each set of 14 words you will be asked to estimate the percentage of words understood. BE CAREFUL, you are not being asked to score you answer sheet but rather to estimate the percentage of words you feel confident you understood correctly. A scale in 7.5% increments will be provided you. Circle the percentage most appropriate.

WORDS

1.__________

2.__________

3.__________

4.__________

5.__________

6.__________

7.__________

8.__________

9.__________

10.__________

11.__________

12.__________

13.__________

14.__________

Estimate in %

```
|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```

b. Sentences

You will be presented with two sets of four sentences. After each sentence you are to identify the last word of the sentence in the blank space provided. As with the words, guessing is encouraged. If you cannot recognize a word, draw a line thru that item. At the end of each set of four sentences you will be asked to estimate the percentage of all of the words in each of the four sentences that you understood. Note, you are being asked to estimate the percentage based on all of the words in each of the four sentences.

1.__________ 2.__________ 3.__________ 4.__________

Estimate in %

¦----¦----¦----¦----¦----¦----¦----¦----¦----¦----¦----¦----¦----¦--¦
0 7.5 15 22.5 30 37.5 45 52.5 60 67.5 75 82.5 90 97.5 100

1.__________ 2.__________ 3.__________ 4.__________

Estimate in %

¦----¦----¦----¦----¦----¦----¦----¦----¦----¦----¦----¦----¦----¦--¦
0 7.5 15 22.5 30 37.5 45 52.5 60 67.5 75 82.5 90 97.5 100

c. Free Running Speech

You will be presented with two 20-25 second passages. The subject matter will be common objects, animals, household plants and food. At the end of each passage you will be asked to make an estimate of the percentage of words which you understood. Be sure to listen to the whole passage and base your estimate on the full 20-25 seconds. A scale in 7.5% increments is provided. Please circle the percentage which is most appropriate and then write to one side of the estimate one word which describes what the passage was about.

Estimate in %

```
|----|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```

Estimate in %

```
|----|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```

Part two of this session will now begin. If you are unsure of any of the above procedures, please ask questions now.

Words

You will be presented with eleven sets of 14 words. Each word will be preceeded by the phrase "will you write". In each set you will be provided with time to identify each individual word. Guessing is encouraged. If you cannot recognize a word, please draw a line thru that item. At the end of each set of 14 words you will be asked to estimate the percentage of words understood. BE CAREFUL, you are not being asked to score your answer sheet but rather to estimate the percentage of words you feel confident you understood correctly. A scale in 7.5% increments will be provided you. Circle the percentage most appropriate.

WORDS

1.__________	8.__________
2.__________	9.__________
3.__________	10.__________
4.__________	11.__________
5.__________	12.__________
6.__________	13.__________
7.__________	14.__________

Estimate in %

|----|----|----|----|----|----|----|----|----|----|----|----|----|--|

0 7.5 15 22.5 30 37.5 45 52.5 60 67.5 75 82.5 90 97.5 100

Sentences

You will be presented with eleven sets of four sentences. After each sentence you are to identify the last word of the sentence in the blank space provided. As with the words, guesing is encouraged. If you cannot recognize a word, then draw a line thru that item. At the end of each set of four sentences you will be asked to estimate the percentage of all of the words in each of the four sentences that you understood. A scale in 7.5% increments will be provided for you. Circle the percentage most appropriate. NOTE, you are being asked to estimate the percentage based on all of the words in in each of the four sentences.

1. __________ 2. __________ 3. __________ 4. __________

Estimate in %

```
|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```

1. __________ 2. __________ 3. __________ 4. __________

Estimate in %

```
|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```

1. __________ 2. __________ 3. __________ 4. __________

Estimate in %

```
|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```

1. __________ 2. __________ 3. __________ 4. __________

Estimate in %

```
|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```

1. __________ 2. __________ 3. __________ 4. __________

Estimate in %

```
|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```

1. __________ 2. __________ 3. __________ 4. __________

Estimate in %

```
|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```

Free Running Speech

You will be presented with eleven 20-25 second passages. The subject matter will be common objects, animals, household plants and food. At the end of each passage you will be asked to make an estimate of the percentage of words which you understood. Be sure to listen to the whole passage and base your estimate on the full 20-25 seconds. A scale in 7.5% increments is provided. Please circle the percentage which is most appropriate and then write to one side of the estimate one word which describes what the passage was about.

```
                          Estimate in %

|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100

                          Estimate in %

|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100

                          Estimate in %

|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100

                          Estimate in %

|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100

                          Estimate in %

|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100

                          Estimate in %

|----|----|----|----|----|----|----|----|----|----|----|----|----|--|
0   7.5  15  22.5  30  37.5  45  52.5  60  67.5  75  82.5  90  97.5 100
```